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VoIP-111A/VoIP-120A

SIP VoIP ATA Adapter

User's Manual

www.airlive.com

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- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected. Ÿ
- Consult the dealer or an experienced radio/TV technician for help.

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Table of Contents

1. Introduction 1

1.1 Introduction 1

1.2 Firmware Upgrade and Tech Support 1

1.3 Features 2

1.4 Overview 2

1.4.1 Hardware 2

1.4.2 Software 3

2. Keypad Interface from Analog Phone Set of ATA 5

3. Configuring the SIP VoIP ATA Adapter 9

3.1 Login Web Management 9

3.1.1 Default Setting 10

3.1.2 System Status 10

3.2 Phone 11

3.2.1 Phone Book 11

3.2.2 Dial Plan 13

3.2.3 Call Services 16

3.2.4 General 23

3.2.5 Volume 24

3.3 Network 24

3.3.1 WAN 24

3.3.2 DDNS 25

3.3.3 VLAN 27

3.3.4 VPN 28

3.3.5 NTP 29

3.4 NAT 30

3.4.1 LAN 30

3.4.2 DMZ & MAC Clone 31

3.4.3 Virtual Server 31

3.5 SIP 33

3.5.1 Service 33

3.5.2 Codes 36

3.5.3 Advanced 37

3.5.4 STUN 44

3.6 Management (Advanced Configuration)..... 45
3.6.1 View Log 46
3.6.2 Auto Provision 47
3.6.3 Auto Update 48
3.6.4 Firmware 50
3.6.5 Advanced 50
3.6.6 Password 53
3.6.7 Tone 55
3.6.8 Default.....58
3.6.9 Language 59
3.7 Save & Reboot 59
3.7.1 Save & Reboot 59
3.8 Logout 60
3.8.1 Logout 60

1 Introduction

1.1 Introduction

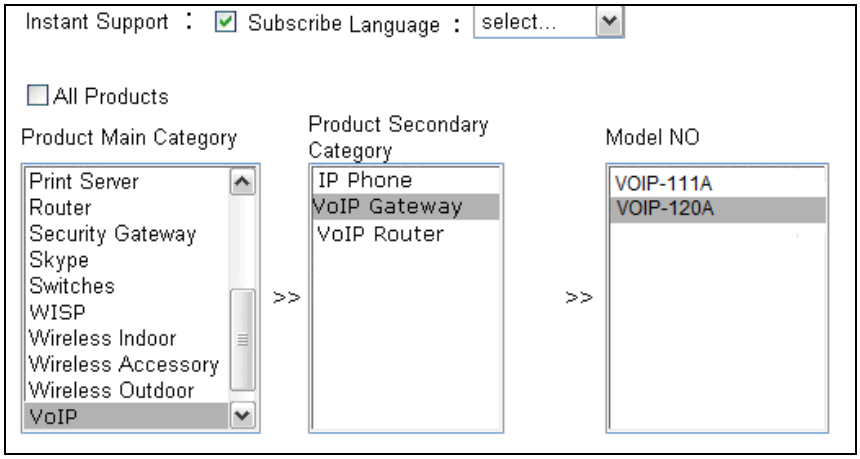
This user's manual is for VoIP-111A (1PSTN+1Phone) / VoIP-120A (2Phone) terminal adapter (ATA). This user's manual explains the IVR instruction, web configuration, and command line configuration for the ATA. Before using the ATA, some setup processes are required to make the ATA work properly. Please refer to the "Instruction of Web Environment" for further information.

1.2 Firmware Upgrade and Tech Support

If you encounter a technical issue that cannot be resolved by information on this guide, we recommend that you visit our comprehensive website support at www.airlive.com. The tech support FAQ are frequently updated with latest information.

In addition, you might find new firmware that either increase software functions or provide bug fixes for VOIP-111A/VOIP-120A. You can reach our on-line support center at the following link:
http://www.airlive.com/support/support_1.php

Since 2009, AirLive has added the “Newsletter Instant Support System” on our website. AirLive Newsletter subscribers receives instant email notifications when there are new download or tech support FAQ updates for their subscribed airlive models. To become an AirLive newsletter member, please visit: http://www.airlive.com/member/member_3.php



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1.3 Features

- Advance SIP VoIP ATA Adapter
- Convert your analog phone into IP Phone
- 1 FXS + 1 FXO ports support 1 IP-Phone with PSTN backup (For VOIP-111A)
- 2 FXS Ports Support 2 IP Phone calls(For VOIP-120A)
- 2 Concurrent calls for 3-Way conference
- FXS port support Phone setting via voice prompt

1.4 Overview

1.4.1 Hardware

The VOIP-111A/VOIP-120A has the following interfaces for Network, telephone interface, LED indication, and power connector.

Ÿ **Two RJ-45 Network interface**

These two interfaces support 10/100Mps Fast Ethernet. You can connect WAN RJ-45 Fast Ethernet port to the ADSL or Switch, and connect the LAN port to your computer.

Ÿ **Two RJ-11 analog telephone jacks interfaces**

You can connect one analog telephone to the terminal adapter and one PSTN line (VOIP-111A). Or, two analog telephone sets at VOIP-120A.

Ÿ **LED Indication**

There are three LED indicators on the ATA to show the Power, Register, and Off-Hook status.

1.4.2 Software

Network Protocol	**Tone**
· SIP v1 (RFC2543), v2 (RFC3261) · IP/TCP/UDP/RTP/RTCP · IP/ICMP/ARP/RARP/SNTP · TFTP Client/DHCP Client/ PPPoE Client · Telnet/HTTP Server · DNS Client · NAT/DHCP Server	· Ring Tone · Ring Back Tone · Dial Tone · Busy Tone · Programming Tone
Codec	**Phone Function**
· G.711: 64k bit/s (PCM) · G.726: 16k / 24k / 32k / 40k bit/s (ADPCM) · G.729A: 8k bit/s (CS-ACELP) · G.729B: adds VAD & CNG to G.729 · G.723.1	· Volume Adjustment · Speed dial key · Phone book · Flash
Voice Quality	**IP Assignment**
· VAD: Voice activity detection · CNG: Comfortable noise generator · LEC: Line echo canceller · Packet Loss Compensation · Adaptive Jitter Buffer	· Static IP · DHCP · PPPoE
	Security
	· HTTP 1.1 basic/digest authentication for Web setup · MD5 for SIP authentication (RFC2069/ RFC 2617)
	QoS
Call Function	· ToS field
· Call Hold · Call Waiting · Call Forward · Caller ID · 3-way conference	**NAT Traversal**
	· STUN
	Configuration
DTMF Function	· Web Browser · Telnet · IVR/Keypad

· In-Band DTMF · Out-of Band DTMF · SIP Info	
·	**Firmware Upgrade**
SIP Server	
· Registrar Server (Five SIP accounts) · Outbound Proxy	· TFTP · HTTP

2 Keypad Interface from Analog Phone Set of ATA

You can use analog phone set's keypad to operate, configure and listen to configuration (IVR play voice messages in English) at ATA without using web interface. The following table is the access code of each feature. Off-Hook analog phone and dial IVR access code and follow the voice prompts to configure ATA IP address and other features.

Group	IVR Action	IVR access code	Parameter(s)	Notes
Function	Dial out from PSTN Line	**0***	None	Press 0* can route your call to PSTN Line from analog phone set directly, you can dial out from PSTN Line. **(For model VOIP-111A only)**
Function	Unlock keypad setting	**#190#**	None	After you unlock keypad setting, you may start to configure ATA from keypad.
Function	Reboot	**#195#**	None	After you hear "Option Successful" from IVR message, please hang-up. The system will reboot automatically.
Function	Factory Reset	**#198#**	None	System reboot automatically. WARNING: ALL "User-Changeable" NONDEFAULT SETTINGS WILL BE LOST including network and service provider data.
Function	Enable PPTP client	**#116#**	None	System will automatically reboot and PPTP client will be enabled
Function	Disable PPTP client	**#117#**	None	System will automatically reboot and PPTP client wll be disabled
Function	Enable VLAN	**#118#**	None	System will automatically reboot and VLAN will be enabled.
Function	Disable VLAN	**#119#**	None	System will automatically reboot and VLAN will be disabled

Function	Enable Call Waiting	**#138#**	None	System will automatically reboot and Call Waiting will be enabled.
Function	Disable Call Waiting	**#139#**	None	System will automatically reboot and Call Waiting will be disabled.
Function	Enable Anonymous	**#140#**	None	System will automatically reboot and Send Anonymous CID was enabled.
Function	Disable Anonymous	**#141#**	None	System will automatically reboot and Send Anonymous CID was disabled.
Function	Blind Transfer	**#510#**	None	This feature was only performed during a phone call.
Function	Attendant Transfer	**#511#**	None	Only be performed in a phone call conversation.
Function	3-way calling (IP Conference)	**#512#**	None	Only be performed in a phone call conversation.
Function	Attendant Transfer	**#514#**	None	Only be performed in a phone call conversation.
Info	Check WAN IP Address	**#126#**	None	IVR will announce the current WAN IP address of the ATA
Info	Check LAN IP Address	**#120#**	None	IVR will announce the current LAN IP address of the ATA
Info	Check IP Type	**#121#**	None	IVR will announce if DHCP in enabled or disabled.
Info	Check the Phone Number	**#122#**	None	IVR will announce current in use VoIP number
Info	Check Network Mask	**#123#**	None	IVR will announce the current network mask of the ATA.
Info	Check Gateway IP Address	**#124#**	None	IVR will announce the current gateway IP address of the ATA.
Info	Check Primary DNS Server Setting	**#125#**	None	IVR will announce the current setting in the Primary DNS field.
Info	Check Firmware Version	**#128#**	None	IVR will announce the version of the firmware running on the ATA.
Setting	Set DHCP client	**#111#**	None	The system will change to DHCP Client type

Setting	Set Static IP Address	**#112xxx*xxx*xxx*xxx#**	Enter IP address using numbers on the telephone keypad. Use the * (star) key when entering a decimal point.	DHCP will be disabled and system will change to the Static IP type.
Setting	Set Network Mask	**#113xxx*xxx*xxx*xxx#**	Enter value-using numbers on the telephone keypad. Use the * (star) key when entering a decimal point.	Must set Static IP first.
Setting	Set Gateway IP Address	**#114xxx*xxx*xxx*xxx#**	Enter IP address using numbers on the telephone keypad. Use the * (star) key when entering a decimal point.	Must set Static IP first.
Setting	Set Primary DNS Server	**#115xxx*xxx*xxx*xxx#**	Enter IP address using numbers on the telephone keypad. Use the * (star) key when entering a decimal point.	Must set Static IP first.
Setting	Set Codec	**#130+[1-8]#**	1:G.711 u-Law, 2: G.711 a-Law, 3:G.723.1, 4: G.729a, 5: G.726 16K, 6:G.726 24K, 7: G.726 32K, 8: G.726 40K,	You can set the codec you want to the first priority.
Setting	Set Handset Gain	**#131+[00~15]#**	Handset Gain from 0~15	You can set the Handset gain to proper value, default is 6
Setting	Set Handset Volume	**#132+[00~12]#**	Handset Volume from 0~12	You can set the Handset volume to proper value, default is 10

Setting	Set Auto Configuration Mode	**#137X#**	Select the auto configuration mode, in the X field, you can press the following; 0:OFF, 1:TFTP, 2:FTP	You can set the auto configuration method you want, default is off
Setting	Set Auto Configuration For TFTP Server	**#135xxx*xxx*xxx*xxx#**	Enter IP address using numbers on the telephone keypad. Use the * (star) key when entering a decimal point.	Must set auto configuration method to TFTP first
Setting	Set Auto Configuration For FTP Server	**#136xxx*xxx*xxx*xxx#**	Enter IP address using numbers on the telephone keypad. Use the * (star) key when entering a decimal point.	Must set auto configuration method to FTP first

3 Configuring the SIP VoIP ATA Adapter

The VOIP-111A/VOIP-120A provide a built-in web browser as management interface. In this chapter, we will explain VOIP-111A/VOIP-120A's web management and how to get into it.

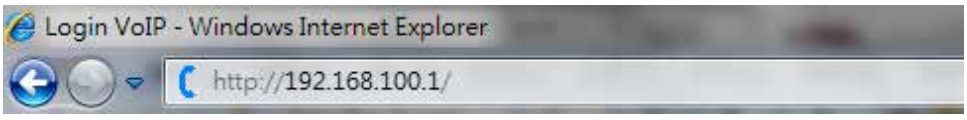
3.1 Login Web Management

The VOIP-111A/VOIP-120A provides a built-in web server. You can use a Web browser to configure the VOIP-111A/VOIP-120A. For more detail instructions, please refer to the procedures below.

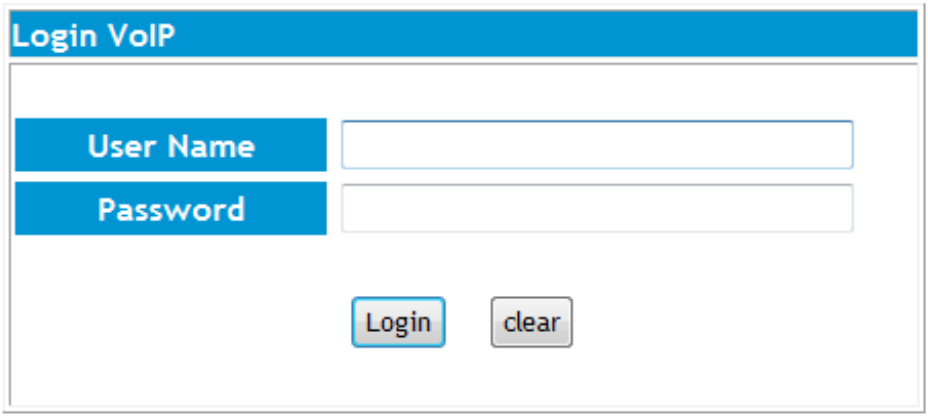
Step 1: Connect a straight-through cable from the VOIP-111A/VOIP-120A's LAN port to a hub or switch while making sure that your computer is also connected to the same hub or switch.

Step 2: Make sure you set your computer's network setting to DHCP mode (obtain IP address automatically). This is done in the Network Connections option that can be accessed in Control Panel.

Step 3: Open your web browser and in the address field, input the following IP address"**http://192.168.100.1"**.



The diagram below will show you an example of the web page menu:



3.1.1 Default Setting

The following information will help you to get start quickly. However, we recommend you to read through the entire manual before you start. Please note the password and SSID are case sensitive.

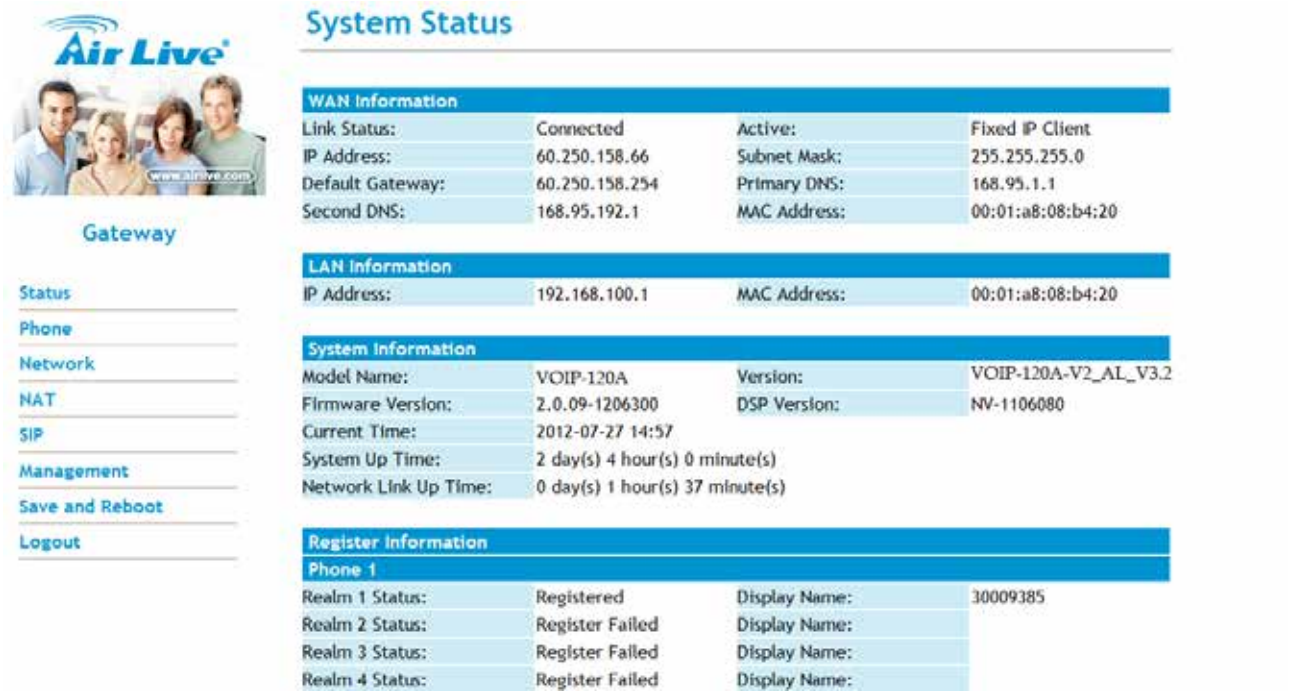
- **Network Mode**: Default NAT Mode
- **WAN Port**: DHCP Client Mode
- **LAN Port**: DHCP Server Enable, IP Address: 192.168.100.1
- **Web Login default link:** http://192.168.100.1
- Account Login :

  Administrator: airlive, Password: airlive
  System: system, Password: airlive
  User: user, Password: airlive

If you change the setting in the Web Management interface, please remember to click "**Submit**" button in that page. After you finished the change of the setting, click the Save function in the left side and click the "**Save**" Button. When you have finished the setting, please click the Reboot function in the left side, and click the "**Reboot**" button in that page. After the system has restarted, all the settings can work properly.

3.1.2 System Status

When you login to the web page, you can see the VOIP-111A /VOIP-120A current system information like firmware version, WAN, LAN, System, and Register…etc in this page. Also you can see the function lists on the left side. You can use the mouse to click the functions you want to set up.



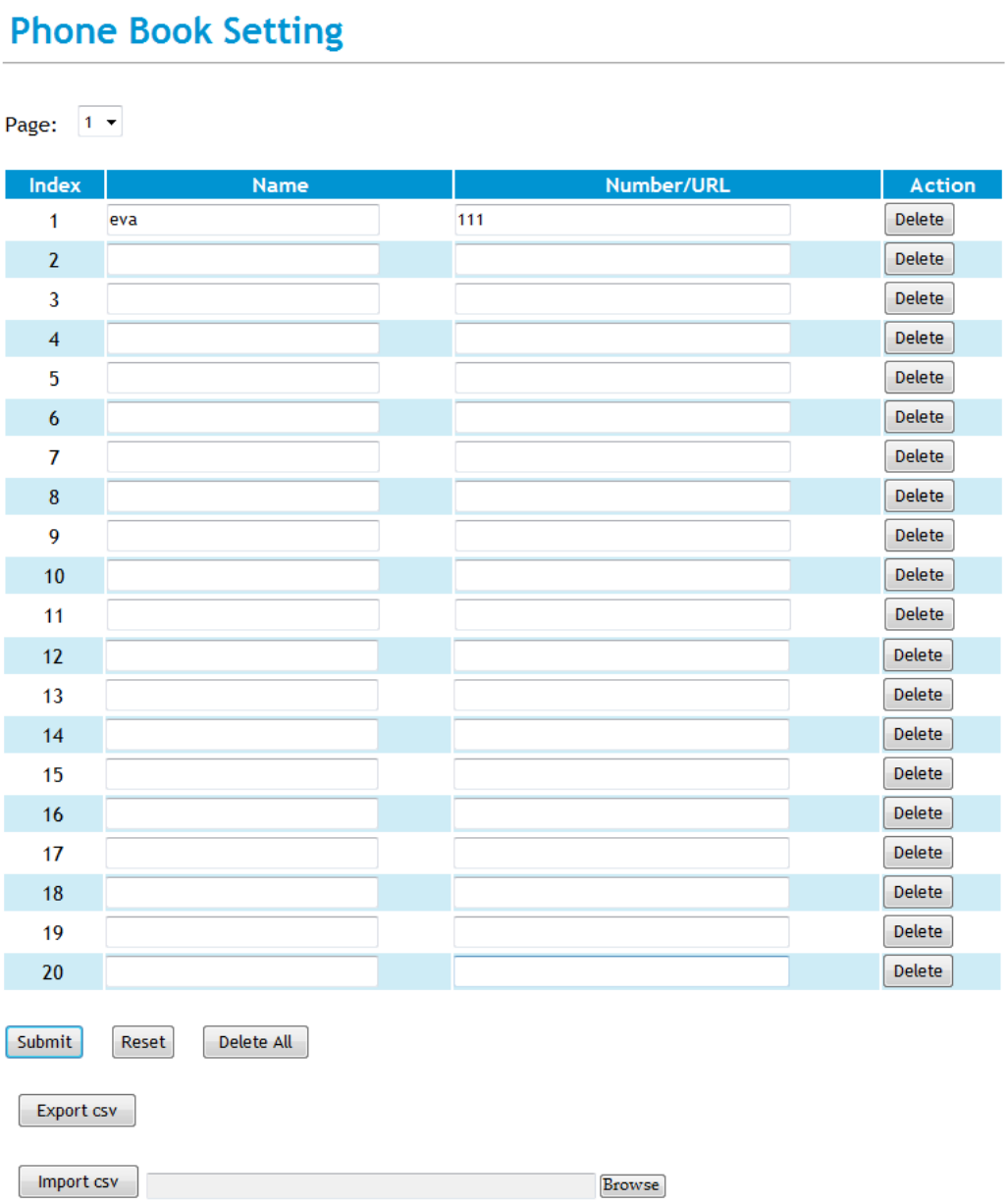
Item	Description
Status	Current device information list
Phone	Phone item provides Phone Book, Dial Plan, Call Service, General setting, Volume settings
Network	Network item provides WAN, DDNS, VLAN, VPN (PPTP/L2TP), NTP (time sync) settings.
NAT	NAT item provides LAN setting, DMZ & Mac Clone, Virtual Server settings.
SIP	SIP item provides Service, Codec, Advanced, STUN settings.
Management	Management item provides Status Log , Auto Config , Auto Update , New Firmware , Advanced , Password , Tones), Default (reset to default), Language settings.
Save & Reboot	Save and Reboot.
Logout	Logout system.

3.2 Phone

This item provides Phone Book, Dial Plan, Call Service, General and volume configuration.

3.2.1 Phone Book

Phone Book provides 140 entries. When you pick up the phone and dial a TEL NO, ATA will compare the Phone Number with the phone book name first. If it matched, ATA will follow the phone book name's number setting to dial. If NOT, the entered number will be dialed out directly.



Item	Description
Page	Default setting is 1. Select the page from Page1 to Page14.
Index	Show index number with 140 entries in total from Phone 0 to 139.
Name	It is speed dial number in speed dial mode. The name field only supports numeric number only such as 0-9. The maximum length is 31 digits.
Number	Dial TEL Number information. Maximum length is 63 digits. For example: 0212345678, 0800024365 or www.dyndns.info.
Action	provide [Delete] button to clear this record's data
Submit [button]	Save data.
Reset [button]	Reset all data.
Delete All [button]	Delete all data.

Example:

In [Phone Book Setting] page,
Set [Index 1], Name: 301, URL: 301@192.168.1.2 .
Set [Index 2], Name: 206, Number: 1747643364.
Set [Index 3], Name: test, Number: 8123478944566.

Index	Name	Number/URL	Action
1	301	301@192.168.1.2	Delete
2	206	1747643364	Delete
3	test	8123478944566	Delete
4			Delete

EX1: pick up phone and dial "301#", in [Index 1] find the name was matched [301], so ATA will use [Number] -> [301@192.168.1.2] to dial out.

EX2: when you pick up phone and dial "206#", ATA will follow the phone book setting to dial 1747643364.

EX3: [Index 3], Name: test, the name is not number type, so it can't be used for speed dialing.

3.2.2 Dial Plan

Provide dial rule and define proxy server prefix code.

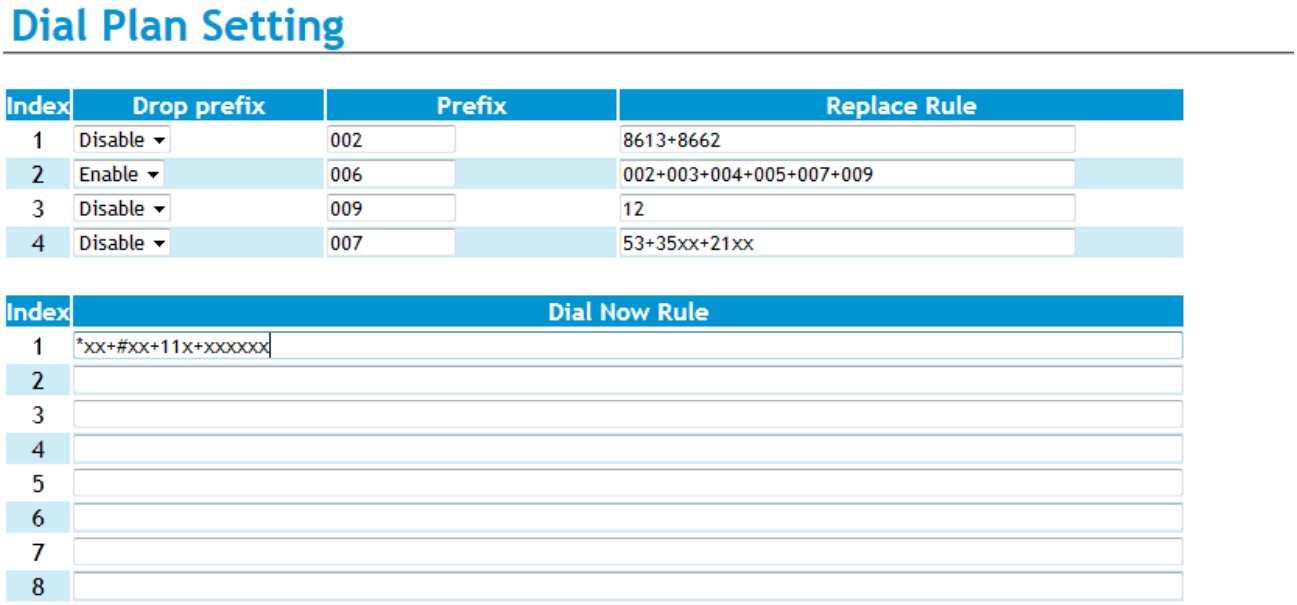
Field	Description
Index	Index number. There are 4 entries to configure ADD and Replace dial codes.
Drop Prefix	Default setting is Disable (It is also an Add digits feature). When it was set to Enable (It is also an Replacement digits feature) and the number match the "Dial Rule" number, ATA will replace the "Rule" field numbers and use "Prefix" field number instead. Ÿ Disable: Add prefix number in front of the called number when called number match "Rule" number. Ÿ Enable: Replace "Rule" number with "Prefix" number when called number match "Rule" number.
Prefix	Added or Replace number. It only allows entering number. Maximum data length is 8 digits.
Rule	Define number manipulation rule. It can be numbers or signs (+, x). The (+) means "Or". The (x) means any numbers which are from 0 to 9. Maximum data length is 40 digits. ***Note:** The first digit can't be 0 if it is 2 digits number length.
Index	Index number. There are 8 entries to configure Dial Out immediately.

Dial Now Rule	Automatic dialing. When the dialing rule match contents in this row, the automatic dialing function will be executed without waiting for "press #" and “Auto Dial Time” to dial out. Both Numbers and symbols can be entered. The number length is 80 digits. Symbols: It was allowed to enter *, #, + and x. +: represents "or". x: any number between 0 to 9. ***Note:** 1st digit cannot be set to "0", because "0" does not determine as the Dial Now Rule. If the Dial Now is set to 0xxxx, the system will not follow the dialing rule to dial out.
Realm 1 prefix	Default setting is 1*. When you dial 1* + called number, ATA will switch to the first account and dial out the called number. Maximum data length is 7 digits.
Realm 2 prefix	Default setting is 2*. When you dial 2* + called number, ATA will switch to the second account and dial out the called number. Maximum data length is 7 digits. ***Note:** If account registers to SIP Server fail, it will not be switched.
Realm 3 prefix	Default setting is 3*. When you dial 3* + called number, ATA will switch to the third account and dial out the called number. Maximum data length is 7 digits. ***Note:** If account registers to SIP Server fail, it will not be switched.
Realm 4 prefix	Default setting is 4*. When you dial 4* + called number, ATA will switch to the fourth account and dial out the called number. Maximum data length is 7 digits. ***Note:** If account registers to SIP server fail, it will not be switched.
Realm 5 prefix	Default setting is 5*. When you dial 5* + called number, ATA will switch to the fifth account and dial out the called number. Maximum data length is 7 digits. ***Note:** If account registers to SIP Server fail, it will not be switched.
Auto Dial Time	Default is 5 seconds; the configuration range is 3 to 9 seconds. ATA will dial out automatically when user didn’t enter any digit within this time period.
Use # as send key	Default is Enable. Define [#] as end of dialing key. Enable: When ATA received [#], it will dial out immediately (no need wait for the auto dial out time). Disable: follow [Auto Dial Time] time to dial out.
Auto PSTN backup	Default is Disable. Provide PSTN auto backup function. When SIP account register to SIP Server fail, FXS port was switched to PSTN line automatically if ATA was set to enable. In the meantime, the FXS port’s dial tone was generated from PSTN line. ***Note:** Please make sure the PSTN line connect to ATA correctly if you want to enable this function.
PSTN feature Code	Default is 0*. This code is to switch the route to PSTN port manually. When you dial 0* you will hear dial tone from PSTN line

	and the call will dial out through PSTN line. Maximum length is 7 digits. It only allows to enter number, * and # digits.
Submit	Save the configuration.
Reset	Clear the configuration.

Example 1: Drop Prefix and Dial Now function.

Step 1: In [Dial Plan Setting] page, the configuration is [Index: 1, Drop prefix: Disable, Prefix: 002, Rule: 8613+8662; Index: 2, Drop prefix: Enable, Prefix: 006, Rule: 002+003+004+005+007+009; Index: 3, Drop prefix: Disable, Prefix: Replace: 009, Rule: 12; Index: 4, Drop prefix: Disable, Prefix: 007, Rule: 53+35xx +21xx; Index: 1, Dial Now Rule: *xx +#xx+11x +xxxxxxxx] (figure1) •



Instruction 1: When the dialing number is like [8613xxxx], it matched [Rule] -> [8613], so ATA will add [prefix] [002] in front of [8613]. The actual dialing number will be [002+8613+xxx].
When the dialing number is like [8662xxxx], it matched [Rule] -> [8662], so ATA will add [prefix] [002] in front of [8662]. The actual dialing number will be [002+8662+xxx].

Instruction 2: When the dialing number is like [002+86xxxx], it matched [Rule] -> [002], so ATA will replace [002] with [Prefix] [006]. The actual dialing number will be [006+86xxxx].
When the dialing number is like [003+77xxxx], it matched [Rule] -> [003], so ATA will replace [003] with [Prefix] [006]. The actual dialing number will be [006+77xxxx].

Instruction 3: Drop prefix: Disable, Replace rule: 009, Rule: 12.

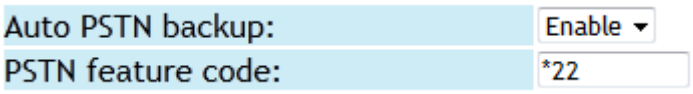
When the dialing number is like [12xxxx], it matched [Rule] -> [12], ATA will add [Prefix] [009] in front of [12]. The actual dialing number will be [009+12xxxx].

Instruction 4: When the dialing number is [53789], it matched [Rule] -> [53], ATA will add [Prefix] [007] in front of [53]. The actual dialing number will be [007+53789].
When the dialing number is [3507], it matched [Rule] -> [35xx], ATA will add [Prefix] [007] in front of [3507]. The actual dialing number will be [007+3507].
When the dialing number is [2199], it matched [Rule] -> [21xx], ATA will add [Prefix] [007] in front of [2199]. The actual dialing number will be [007+2199] •

Instruction 5: When the dialing number is [*00, *01, *02… *99], it matched [Dial Now Rule] -> [*xx]. ATA will dial out immediately.
When the dialing number is [#00, #01, #02… #99], it matched [Dial Now Rule] ->[#xx]. ATA will dial out immediately.
When the dialing number is [110, 111, 112 … 119], it matched [Dial Now Rule] -> [11x]. ATA will dial out immediately.
When the dialing number is [123456], it matched [Dial Now Rule] -> [xxxxxx]. ATA will dial out immediately.

Example 2: PSTN feature code function.

Step 1: In [Dial Plan Setting] page, the configuration is [Auto PSTN Backup: Enable, PSTN feature Code: *22].

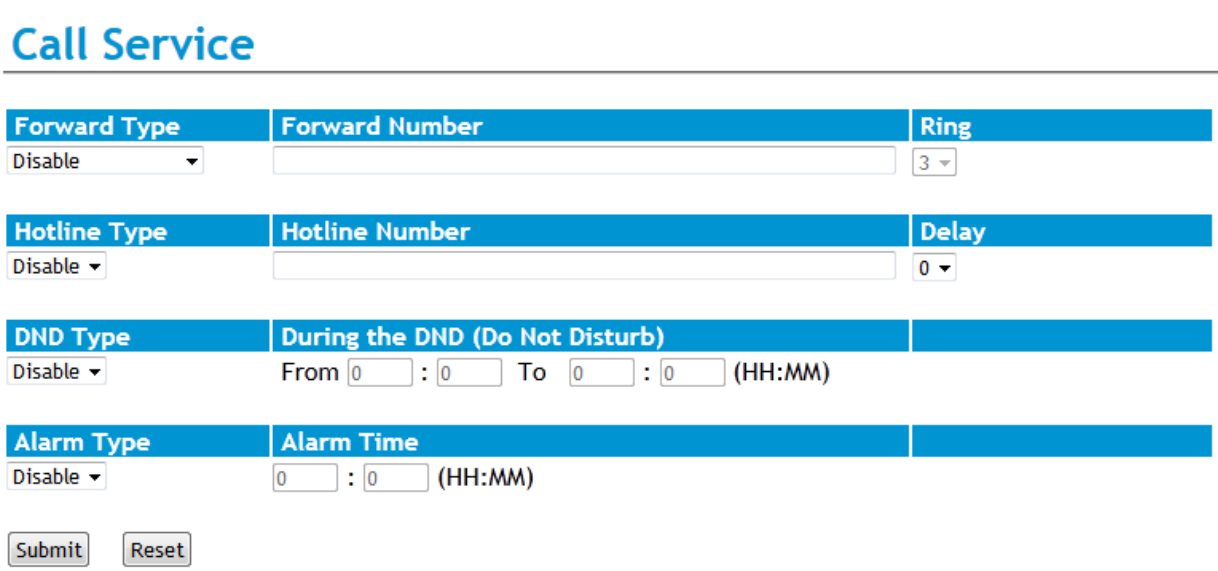


Description 1: When ATA registered to SIP Server fail, its FXS phone line was forced to connect with PSTN line when user is going to make a call. The Dial Tone was provided from PSTN line instead.

Description 2: When ATA registered to SIP Server successfully, press dials code *22 from analog phone set to force ATA switch to PSTN line manually. The Dial Tone was provided from PSTN line instead.

3.2.3 Call Services

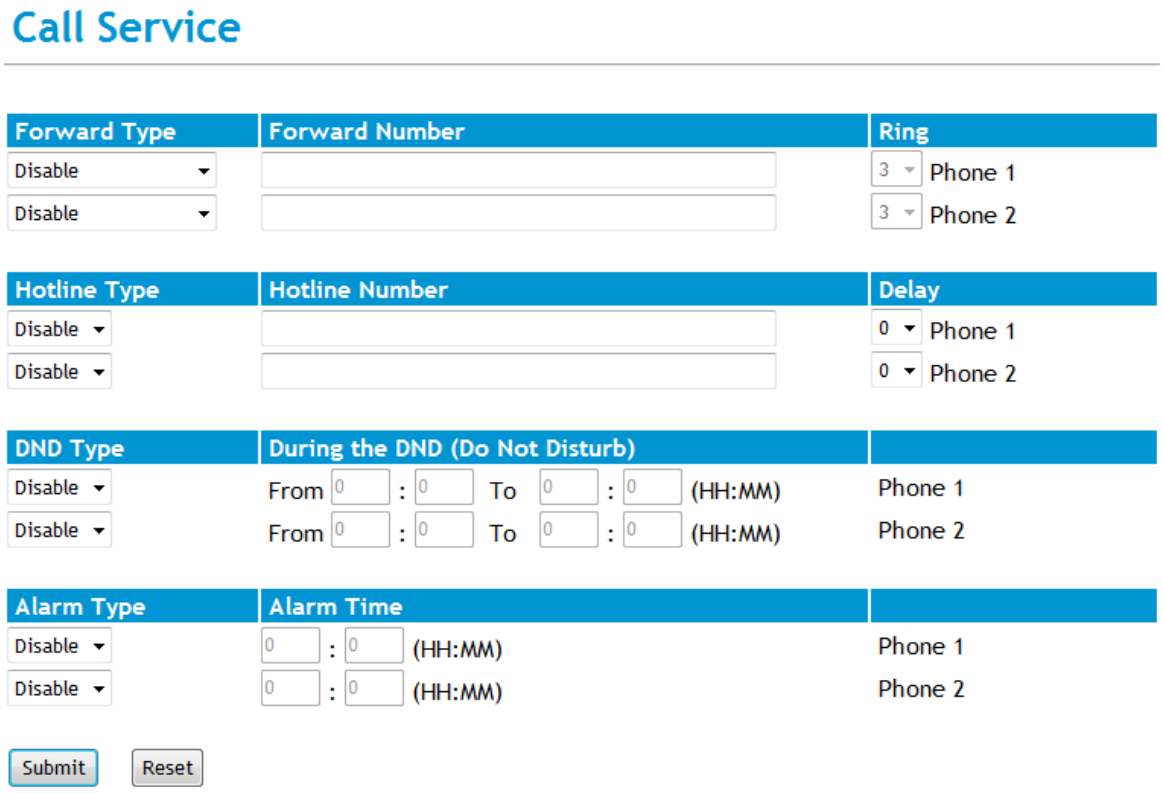
Call Services provides the Forward, Hotline, DND, Alarm functions.



This figure is for VOIP-111A

Field	Description
Forward Type of phone1	Default is Disable. To configure Phone 1 forward type. Here provides 5 options: Disable Always, Busy, No Answer, Busy or No Answer. ***Note:** Please make sure your service provider support this forward function.
Forward Number of phone1	To configure Phone 1 forward number, simply dial [number or digit string]. The maximum digit length is 63.
Rings of phone 1	That feature is used for no answer forward only. Default is 3 rings. When there is no answer after configured rings, ATA forward to pre-configured number automatically. The configuration ring range is 2 to 8 rings. This mode only supports Forward Type: No Answer.
Hotline Type of phone 1	Phone 1 hotline function, default is Disable. If Enable, ATA will dial the hotline number immediately when you pick up phone. ***Note:** You need to configure Hotline number in advance. See the next row.
Hotline Number of phone 1	Configure Phone 1 hotline number. You can enter IP address or number or digit string and the maximum length is 63 digits. For instance, IP address: 192.168.1.23 or telephone number: 0800024365.
Delay of phone 1	When you pick up the phone before dialing, ATA start to count time until the first digit was dialed. Default time is 3 seconds. ATA will use Hotline number to dial if configured time was expired. The configuration range is 1 to 6 seconds.

DND Type of phone 1	Configure Phone 1 DND function, default is Disable. When you set to Enable, ATA will response SIP command 486 message (Busy status) to calling user once an call incoming. There are 3 options to configure: Disable, Always, Period (DND enable according pre-defined time period, refer to time setting at next row).
DND Time of phone 1	This command is to configure DND time period at phone 1. Default is From 0:0(start time) To 0:0(end of time). The time format is 24 hours system (hh/mm, Hours/Minutes). Each field has 2 digits number only.
Alarm Type of phone 1	Default is Disable. Configure Phone 1 alarm function. When you set to enable, phone 1 will Ring according to pre-configured (see next row how to configure alarm time) alarm time. The alarm Ring last for 1 minute. To cancel alarm setting, simply pickup handset and hand up. The default ring time is 1 minute.
Alarm Time for phone 1	Default is 0:0(Hour/Minute). The time format is 24 hours system (hh/mm). Each field allows to enter 2 digits number only.
Submit	Save the configuration.
Reset	Clear the configuration.



This figure is for VOIP-120A

Field	Description
Forward Type of phone1	Default is Disable. To configure Phone 1 forward type. Here provides 5 options: Disable Always, Busy, No Answer, Busy or No Answer. ***Note:** Please make sure your service provider support this forward function.
Forward Number of phone1	To configure Phone 1 forward number, simply dial [number or digit string]. The maximum digit length is 63.
Rings of phone 1	That feature is used for no answer forward only. Default is 3 rings. When there is no answer after configured rings, ATA forward to pre-configured number automatically. The configuration ring range is 1 to 6 rings. This mode only supports Forward Type: No Answer.
Forward Type of phone 2	Default is Disable. To configure Phone 2 forward type. Here provides 5 options: Disable, All (unconditional), Busy, No Answer, Busy or No Answer. ***Note:** Please make sure your service provider support this forward function.
Forward Number of phone 2	To configure Phone 2 forward number, simply dial [number or digit string]. The maximum digit length is 63.
Rings of phone 2	That feature is used for no answer forward only. Default is 3 rings. When there is no answer after configured rings, ATA forward to pre-configured number automatically. The configuration ring range is 1 to 6 rings. This mode only supports Forward Type: No Answer.
Hotline Type of phone 1	Phone 1 hotline function, default is disable. Enable: ATA will dial the hotline number immediately when you pick up phone. ***Note:** You need to configure Hotline number in advance. See the next row.
Hotline Number of phone 1	Configure Phone 1 hotline number. You can enter IP address or number or digit string and the maximum length is 63 digits. For instance, IP address: 192.168.1.23 or telephone number: 0800024365.
Delay of phone 1	When you pick up the phone before dialing, ATA start to count time until the first digit was dialed. Default delay time is 3 seconds. ATA will use Hotline number to dial if configured time was expired. The configuration range is 1 to 6 seconds.
Hotline Type of phone 2	Phone 2 hotline function, default is Disable. Enable: ATA will dial the hotline number immediately when you pick up phone. ***Note: You need to configure Hotline number in advance. See the next row.**
Hotline Number of phone 2	Configure Phone 2 hotline number. You can enter IP address or number or digit string and the maximum length is 63 digits. For instance, IP address: 192.168.1.23 or telephone number: 0800024365.

Delay of phone 2	When you pick up the phone before dialing, ATA start to count time until the first digit was dialed. Default delay time is 3 seconds. ATA will use Hotline number to dial if configured time was expired. The configuration range is 1 to 6 seconds.
DND Type of phone 1	Configure Phone 1 DND function, default is Disable. When you set to Enable, ATA will response SIP command 486 message (Busy status) to calling user once an call incoming. There are 3 options to configure: Disable, Always, Period (DND enable according pre-defined time period, refer to time setting at next row).
DND Time of phone 1	This command is to configure DND time period at phone 1. Default is From 0:0(start time) To 0:0(end of time). The time format is 24 hours system (hh/mm, Hours/Minutes). Each field has 2 digits number only.
DND Type of phone 2	Configure Phone 2 DND function, default is Disable. When you set to Enable, ATA will response SIP command 486 message (Busy status) to calling user once an call incoming. There are 3 options to configure: Disable, Always, Period (DND enable according pre-defined time period, refer to time setting at next row).
DND Time of phone 2	This command is to configure DND time period at phone 2. Default is From 0:0(start time) To 0:0(end of time). The time format is 24 hours system (hh/mm, Hours/Minutes). Each field has 2 digits number only.
Alarm Type of phone 1	Default is Disable. Configure Phone 1 alarm function. When you set to enable, phone 1 will Ring according to pre-configured (see next row how to configure alarm time) alarm time. The alarm Ring last for 1 minute. To cancel alarm setting, simply pickup handset and hand up. The default ring time is 1 minute.
Alarm Time for phone 1	Default is 0:0(Hour/Minute). The time format is 24 hours system (hh/mm). Each field allows to enter 2 digits number only.
Alarm Type of phone 2	Default is Disable. Configure Phone 2 alarm function. When you set to enable, phone 2 will Ring according to pre-configured (see next row how to configure alarm time) alarm time. The alarm Ring last for 1 minute. To cancel alarm setting, simply pickup handset and hand up. The default ring time is 1 minute.
Alarm Time for phone 2	Default is 0:0(Hour/Minute). The time format is 24 hours system (hh/mm). Each field allows entering 2 digits number only.
Submit	Save the configuration.
Reset	Clear the configuration.

Example1: Forward Feature

(1) Always (Unconditional Transfer):

Step1: Go to [Call Service Setting] page and configure [Forward Type: All, Forward Number: 812345678].

Step2: Route an incoming call to desire routes to dial according to configuration contents of [Forward Type] and [Forward Number].

(2) Busy (Busy Transfer):

Step1: Go to [Call Service Setting] page and configure [Forward Type: Busy, Forward Number: 405, Rings: 3].

Step2: Route an incoming call to desire routes to dial according to configuration contents of [Forward Type] and [Forward Number] when ATA is on line (Busy).

(3) No Answer (No Answer Forward):

Step1: Go to [Call Service Setting] page and configure [Forward Type: No Answer, Forward Number: 031237788, Rings: 3].

Step2: Route an incoming call to desire routes to dial according to configuration contents of [Forward Type] and [Forward Number] when ATA has received configured Ring cycles and nobody answer this call.

(4) Busy or No Answer (Busy or No Answer Forward):

Step1: Go to [Call Service Setting] page and configure [Forward Type: Busy or No Answer, Forward Number: 031237788, Rings: 3].

Step2: Route an incoming call to desire routes to dial out according to configuration contents of [Forward Type] and [Forward Number] when ATA has received configured Ring cycles and nobody answer this call or ATA is on line (Busy).

Example 2: Hotline Feature

(1) Dial a hotline number:

Step1: Go to [Call Service Setting] page and configure [Forward Type: Enable, Hot Line number: 82341234, Delay: 3].

Step2: Dial number according to configuration contents of [Hotline Number] when ATA pickup handset and wait for [Delay] time was expired before dialing any digits.

(2) Dial an IP Address:

Step1: Go to [Call Service Setting] page and configure [Hotline Type: Enable, Hot Line number: 192.168.50.4, Delay: 3].

Hotline Type	Hotline Number	Delay
Enable	192.168.50.4	3 Phone 1

Step2: Dial IP address directly according to configuration contents of [Hotline Number] when ATA pickup handset and wait for [Delay time] was expired before dialing any digits.

Example 3: DND feature (Do Not Disturb)

Reject incoming call at assigned time interval:
Step1: Go to [Call Service Setting] page and configure [DND Type: Period, from: 18:15, to: 22:20].

DND Type	During the DND (Do Not Disturb)	
Period	From 18 : 15 To 22 : 20 (HH:MM)	Phone 1

Step2: ATA rejects an incoming call when it happened at assigned time interval according to configuration at [DND Time].

Reject all incoming call:

Step1: Go to [Call Service Setting] page and configure [DND Type: Always].

Hotline Type	Hotline Number	Delay
Enable	192.168.50.4	3 Phone 1

STEP 2: ATA rejects all incoming calls without any exception.

Example 4: Alarm Feature

Reject incoming call at assigned time interval:

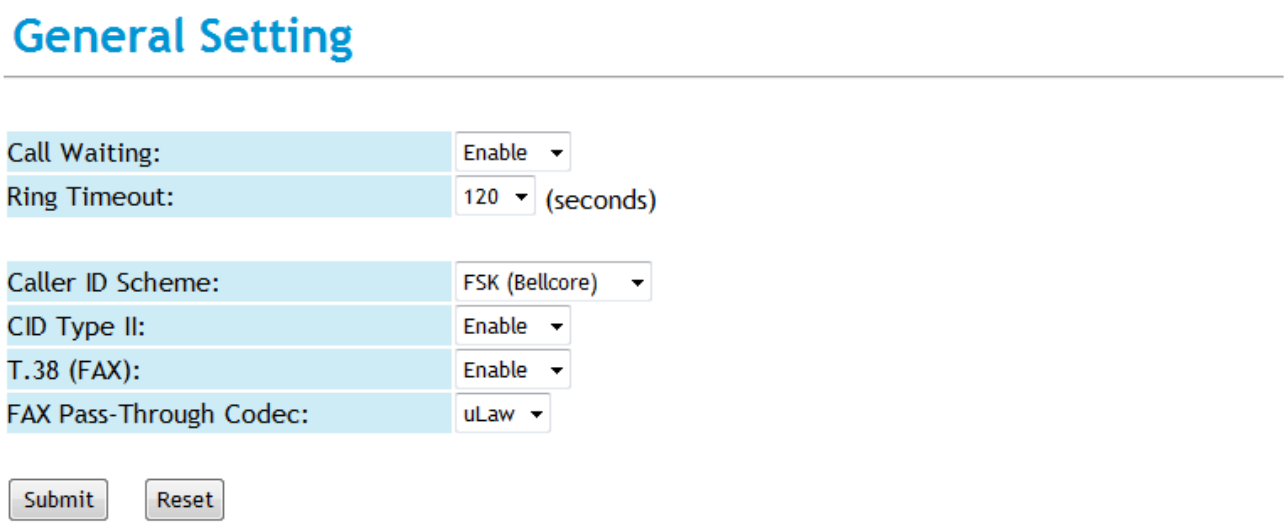
Step1: Go to [Call Service Setting] page and configure [Alarm Type: Enable, Alarm Time: 21:00].

Alarm Type	Alarm Time	
Enable	21 : 0 (HH:MM)	Phone 1

Step2: ATA's analog phone set start Ring when 21:00PM arrive daily and it is lasting for default time (i.e. default time is 1 minute). It stops ring immediately when you pickup handset.

3.2.4 General

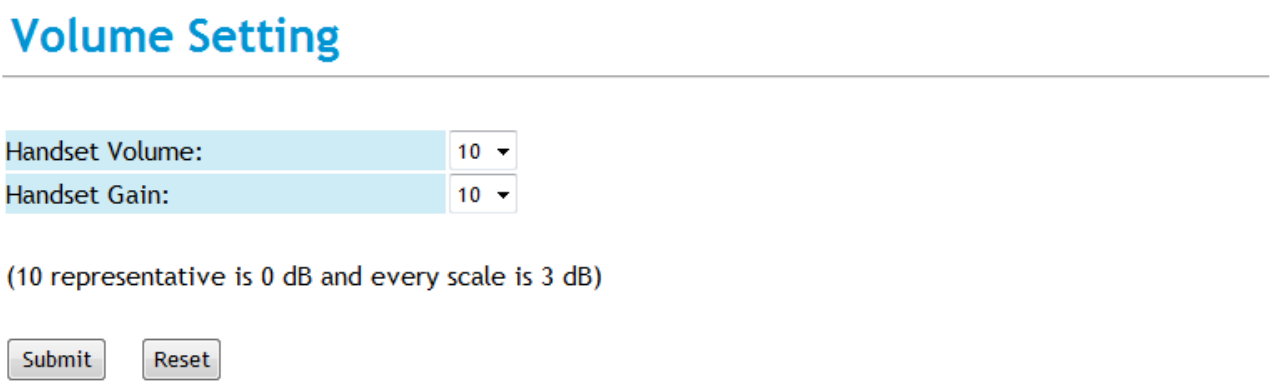
Provide Caller ID, Call waiting, auto answer and T.38 FAX transmission.



Field	Description
Call Waiting	Enable/Disable call waiting function. This feature allows you to answer the incoming call when you are on line at another call. When you are on line to talk, a "Du Du" sound was heard to remind you there is an incoming call. To answer this incoming call, simple activate HOLD feature to hold existing call and answer incoming call.
Ring Timeout	Default setting is 60 seconds. ATA responses Busy tone (SIP command 486) to caller when nobody answer incoming call once configured time was expired. The configured timeout option is: 20, 40, 60, 80, 120, 180 and 240 seconds.
Caller ID Scheme	Default is Disable. The Caller ID supports the following protocol: FSK Bellcore, DTMF (Caller ID before first Ring), CID-Japan, DTMF-Brazil and DTMF-Denmark mode. *Note: Your analog telephone set MUST support proper Caller ID mode to show CID number.
CID Type II	Default is Disable. If ATA has enabled both CID Type II and Call Waiting function, ATA will show the incoming call Caller ID when you are on line (busy). *Note: Your analog telephone set MUST support proper Caller ID Type II mode to show CID number.
T.38 (FAX)	Default is Enable to support T.38 FAX transmission function.
T.38 Pass-through codec	Default codec is G.711 u-Law (enable) to support T.38 FAX passes through. ATA only uses codec either G.711 u-law or G.711 a-law to transmit FAX over T.38 protocol.
Submit [button]	Save the configuration.
Reset [button]	Clear the configuration.

3.2.5 Volume

This function is to adjust volume of microphone and speaker at analog phone set, and FXO port's volume as well.



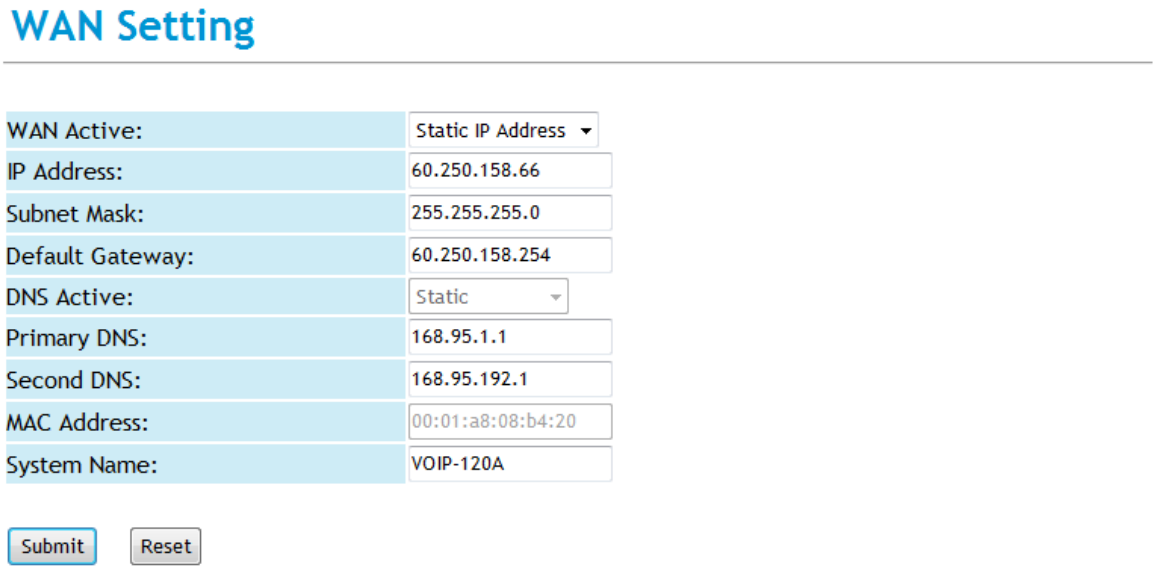
Field	Description
Handset Volume	Default is 10. Control the volume of the Handset receiver from (0 to 14). Maximum length is 2 digits.
Handset Gain	Default is 10. Control the handset gain (microphone volume to send to remote site) from (0 to 15). Maximum length is 2 digits.
Submit [button]	Save the configuration.
Reset [button]	Clear the configuration.

3.3 Network

In Network you can check the Network status, WAN, DDNS, VLAN, VPN and NTP Settings.

3.3.1 WAN

Provide WAN port configuration.

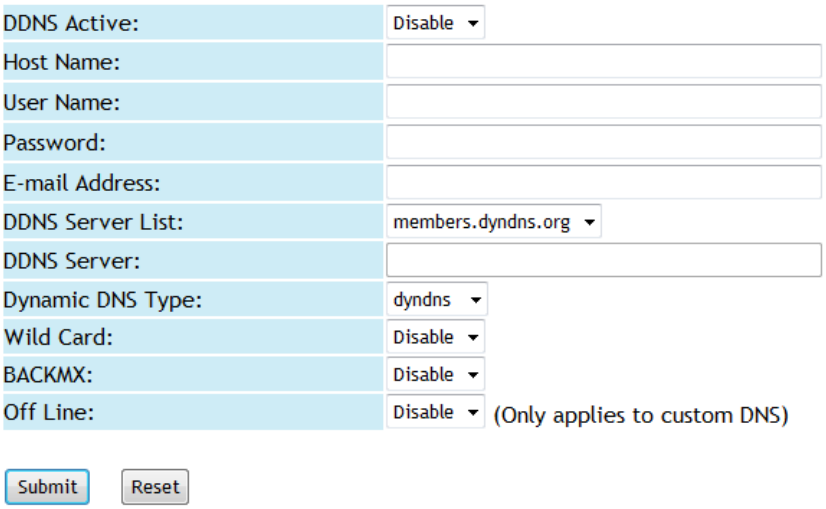


Field	Description
Type	Default is DHCP Client and provides Static IP Address, DHCP (gains IP Address automatically) and PPPoE: ADSL Dialing method.
IP Address	Current IP Address list or any IP Address (IPv4 format). If you would like to change IP Address, please set IP Type as “Static IP” first. Maximum length is 15 digits.
Subnet Mask	Current Subnet list. A **subnetwork**, or **subnet**, is a logically visible subdivision of an IP network. The format is xxx.xxx.xxx.xxx. Maximum length is 15 digits.
Default Gateway	Current default gateway list. Gateway is a node (a router) on a TCP/IP network that serves as an access point to another network. The format is xxx.xxx.xxx.xxx. Maximum length is 15 digits.
DNS Type	Default is Auto. Define DNS server mode to be Fixed and Auto. Fixed: Manual set DNS server. Auto: Network server provides DNS. It only supports in Type: DHCP Client and PPPoE.
DNS Server1	Default: 168.95.192.1. It can be IP address or Domain Name. Format: xxx.xxx.xxx.xxx. Maximum length is 15 digits.
DNS Server2	Default: 168.95.1.1. It can be IP address or Domain Name. Format: xxx.xxx.xxx.xxx. Maximum length is 15 digits.
MAC ID	Display MAC address information.
Host Name	Default: product name. Numbers or strings are both acceptable. Length: 15 digits.
PPPoE User Name	Provides user’s name of PPPoE Server, it can be numbers or strings. Length is 32 digits.
PPPoE Password	Provides password of PPPoE Server, it can be numbers or strings. Length is 32 digits.
PPPoE Service Name	Define Service Name; It can be IP address or Domain Name. Length: 32 digits. *Note: This name is provided from ISP. If you don’t know it, please don’t change anything here.
PPPoE AC Name	Define AC Name; It can be IP address or Domain Name. Length: 32 digits. *Note: This name is provided from ISP. If you don’t know it, please don’t set anything here.
Submit [button]	Save the configuration.
Reset [button]	Clear the configuration.

3.3.2 DDNS

Dynamic DNS provides a residential user's Internet gateway that has a variable, often changing IP address with a well known hostname resolvable through standard DNS queries.

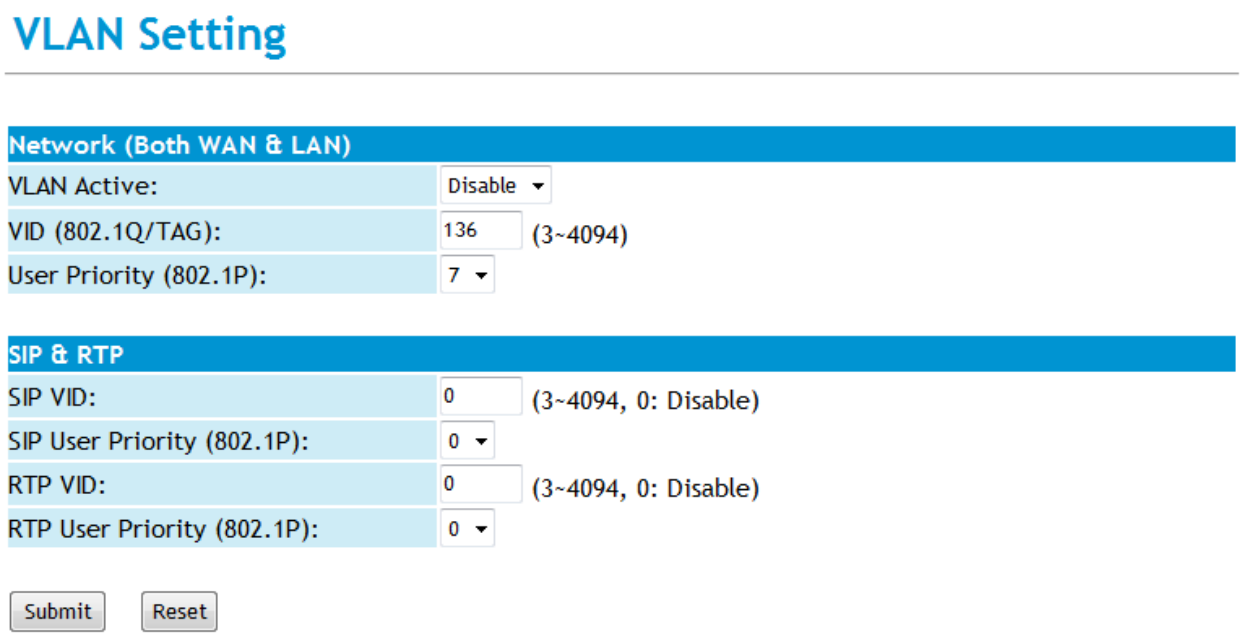
Dynamic DNS Setting



Field	Description
Active	Default is Disable; The DDNS function will be enabled when you set to Enable.
Host name	Enter Host name which can be IP Address or Domain Name. Format: xxx.xxx.xxx.xxx. Length is 63 digits.
User Name	Enter user's name for registering to DDNS Server.
Password	Enter the password. Maximum length is 63 digits.
E-mail address	Enter E-mail address. Maximum length is 63 digits.
DDNS Server List	Default is Disable; Configure your service provider here. Provide option: User input, members.dyndns.rog and www.dtdns.com mode.
DDNS Server	Enter DDNS Server which can be IP Address or Domain Name. Format: xxx.xxx.xxx.xxx. Maximum length is 63 digits.
Type	Default is dyndns. Provide 3 options: dyndns, statdns and customer mode.
Wild Card	Default is Disable. Provides 3 options: Enable, Disable and Nochg. ***Note:** Please make sure your DDNS provider supports this feature when you set it to enable.
BACKMX	Default is Disable; The backup MX function will was activate when you set it to enable. It provides two options: Disable and Enable. ***Note:** Please make sure your DDNS provider supports this feature when you set it to enable.
Off Line	Default is Disable. The Off Line function will be activated when you set it to enable. It provides two options: Disable and Enable. ***Note:** Please make sure your DDNS provider supports this feature when you set it to enable.
Submit [button]	Save the configuration.
Reset [button]	Clear the configuration.

3.3.3 VLAN

Provide Network, SIP and RTP VLAN function. This feature needs to work with VLAN Router.

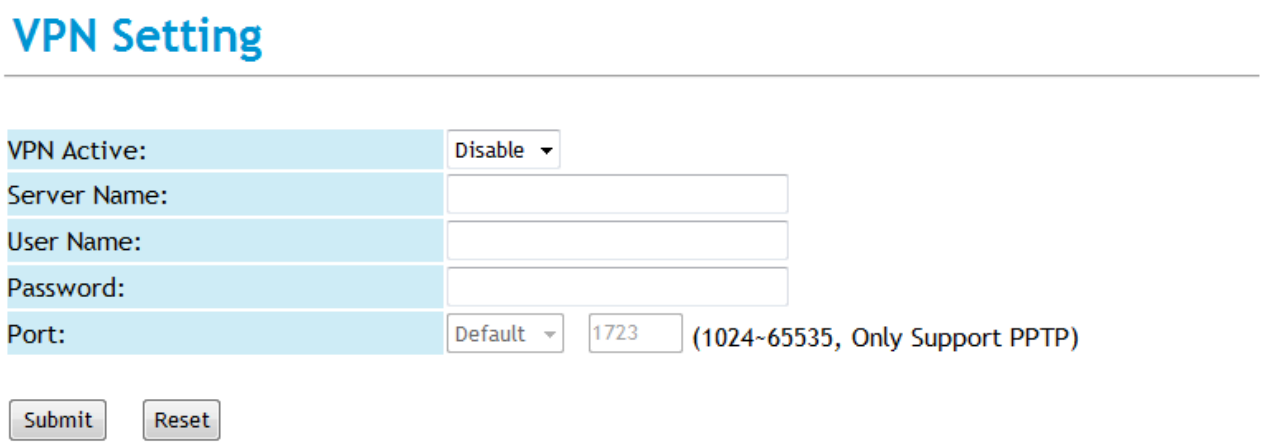


Field	Description
Network (Both WAN & LAN)	Configure VLAN function of both WAN and LAN ports.
VLAN Packets	Default is Disable. ATA supports VLAN function (accept VLAN packets) when you set it to enable.
VID (802.1Q/ TAG)	Default: 136. Provide Virtual LAN ID (VLAN or VID) for VLAN Server. Data range: 3~4097. Maximum length is 4 digits.
User Priority (802.1P)	Default: 0. Set the user's priority. Data range: 0 to 7.
CFI	Default is 0. To set Canonical Format Indicator (CFI) for one byte. Data Range is (0~1). The CFI bit is used to indicate that all MAC addresses present in the MAC data field are in canonical format. This field is interpreted differently depending on whether it is an Ethernet-encoded tag header or a SNAP-encoded tag header. In SNAP-encoded TPID the field indicates the presence or absence of the canonical format of addresses. In Ethernet-encoded TPID, it indicates the presence of the Source-Routing Information (RIF) field after the length field. The RIF field indicates routing on Ethernet frames.
SIP & RTP	Define SIP & RTP VLAN feature.
SIP VID	Default is 0 (disable). This feature is to define SIP VLAN ID. Range is 3~4094. This feature is an independent one which do not need to Enable [VLAN Packets] if you want to enable this feature. Data length is 4 digits.
User Priority (802.1P)	Default is 0 (disable). Define SIP package priority. Range is 0~7.

Field	Description
CFI	Default: 0. To set Canonical Format Indicator (CFI) for one byte. Data Range is (0~1).
RTP VID	Default is 0 (disable). Define SIP VLAN ID. Range is 3~4094. This feature is an independent one which do not need to Enable [VLAN Packets] if you want to enable this feature. Data length is 4 digits.
User Priority (802.1P)	Default is 0 (disable); Define RTP package priority. Range: 0~7
CFI	Default: 0. To set Canonical Format Indicator (CFI) for one byte. Data Range is (0~1)
Submit [button]	Save the configuration.
Reset [button]	Clear the configuration.

3.3.4 VPN

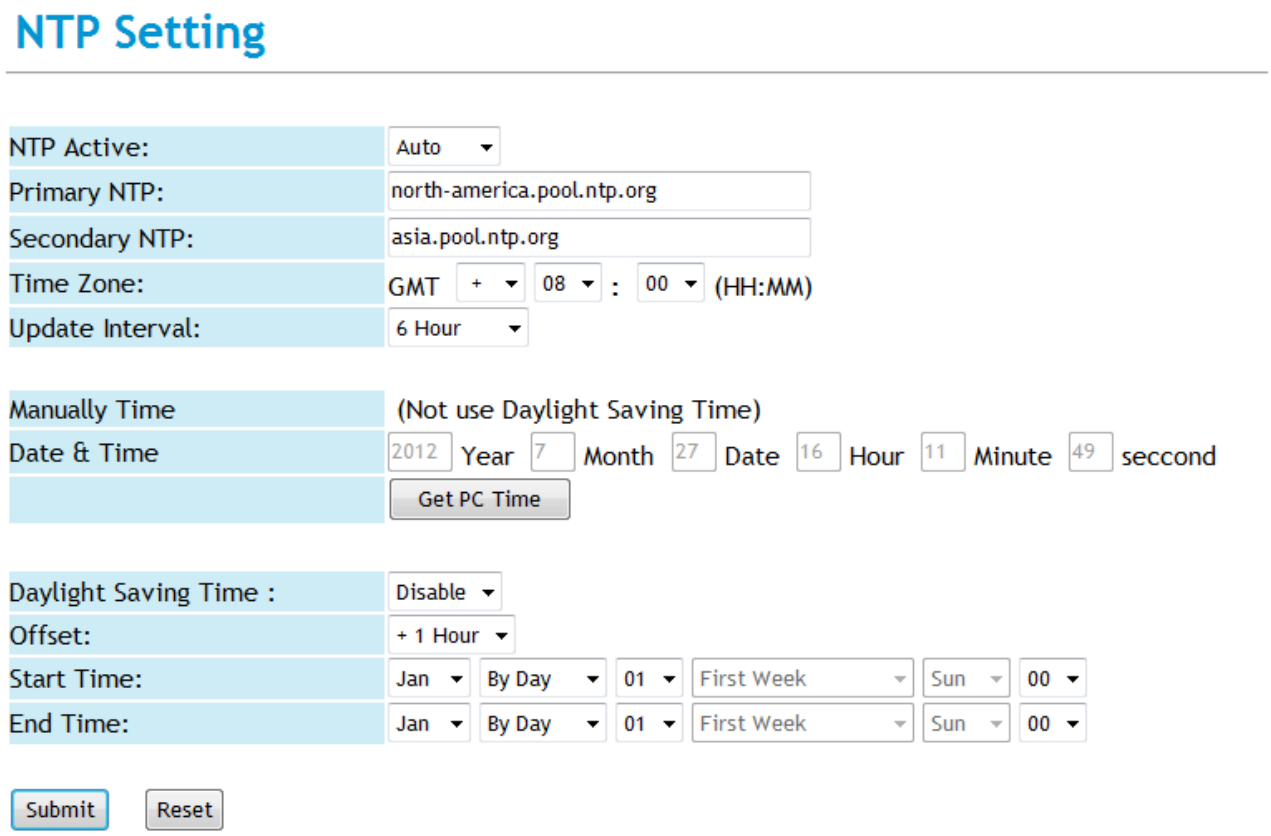
Provide [PPTP or L2TP] configuration. If you want to configure VPN function via webpage, please go to LAN port to do it.



Field	Description
Type	Default is Disable. Provide PPTP/L2TP connection mode.
Server Name	Enter PPTP/L2TP Server information which can be IP Address or Domain Name. Format is: xxx.xxx.xxx.xxx. Maximum length is 63 digits.
User Name	Enter PPTP/L2TP Server user's name or IP address which can be number or strings. Maximum length is 63 digits.
Password	Enter PPTP/L2TP password which can be numbers or strings. Maximum length is 63 digits.
Port Number	Default PPTP Port is 1723. Provide 2 options: Default, Customer (User defines port) and the range is 1024~65535. Only number and length is 5 digits.
Submit [button]	Save the configuration.
Reset [button]	Clear the configuration.

3.3.5 NTP

Provide time synchronization and daylight saving function.


NTP Setting
NTP Active: Auto
Primary NTP: north-america.pool.ntp.org
Secondary NTP: asia.pool.ntp.org
Time Zone: GMT + 08 : 00 (HH:MM)
Update Interval: 6 Hour
Manually Time (Not use Daylight Saving Time)
Date & Time 2012 Year 7 Month 27 Date 16 Hour 11 Minute 49 seccond
Get PC Time
Daylight Saving Time : Disable
Offset: + 1 Hour
Start Time: Jan By Day 01 First Week Sun 00
End Time: Jan By Day 01 First Week Sun 00
Submit
Reset

Field	Description
NTP Active	Default is Auto. This feature is to Auto/Manual Time Sync feature. When it was set to Disable, you may configure Time manually.
Primary NTP	Default is north-america.pool.ntp.org to be Primary time sync server. It can be IP address or Domain Name. The format is xxx.xxx.xxx.xxx. Maximum length is 63 digits.
Secondary NTP	Default is asia.pool.ntp.org to be Secondary time sync server. It can be IP address or Domain name. The format is xxx.xxx.xxx.xxx. Maximum length is 63 digits.
Time Zone	Default is GMT + 08:00 (hh:mm). Provide option is: +/-, 00~13(hour) 00, 15, 30, and 45 (minute).
Update Internal	Default is 24 Hour. To configure the interval time for Time Sync. The options are: 1 min., 5 min., 30 min., 1 hour, 3 hour, 6 hour, 12 hour, and 24 hour.
Daylight Saving Time	Default is Disable. Define the Daylight Saving time. When you set to Enable, time will be adjusted by the Daylight Saving parameter.
DST Offset	Default is +1 Hour. Adjust the Daylight Saving time. Option: -2 hour, -1 hour, +1 hour, +2 hour.
DST Start Date	Configure the Daylight Saving start time. Option: Day of Month, Week of Month.

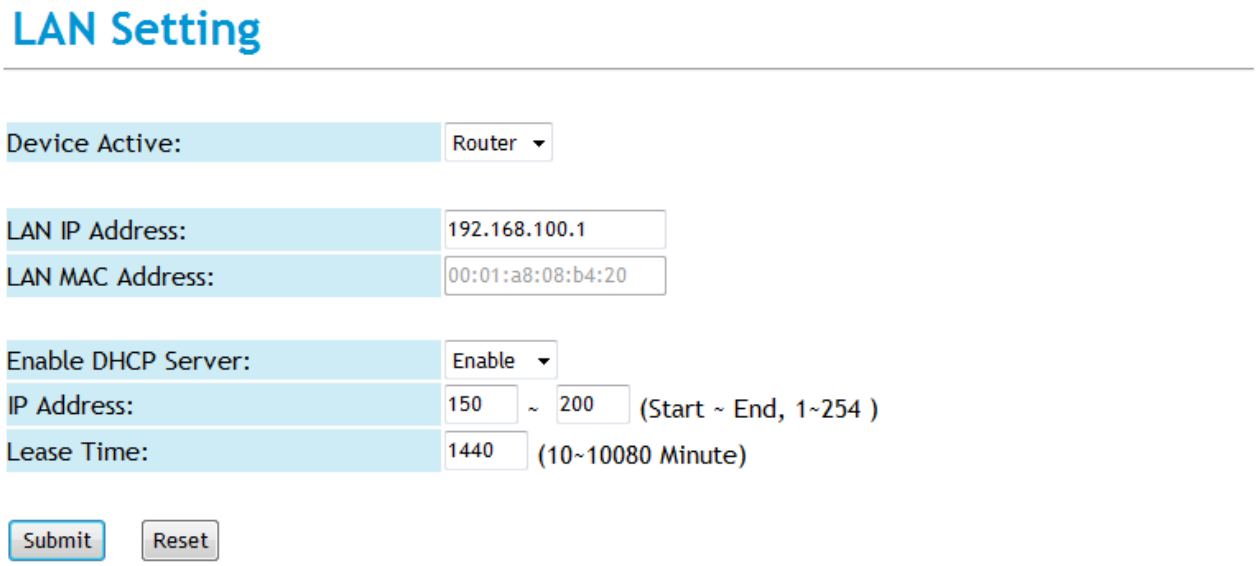
Field	Description
DST End Date	Configure the Daylight Saving start time Option: Day of Month, Week of Month.
Submit [button]	Save the configuration.
Reset [button]	Clear the configuration.

3.4 NAT

Provide LAN, DMZ & MAC Clone and Virtual Server functions.

3.4.1 LAN

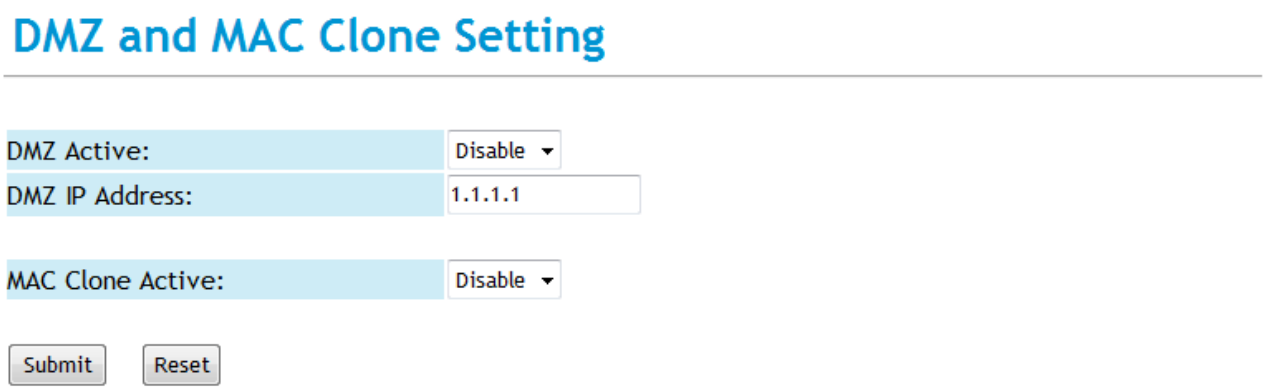
Provide LAN port configuration setting including DHCP server function.



Field	Description
LAN Mode	Default is NAT mode. The option is: Bridge and NAT mode. Bridge: When set to Bridge, WAN and LAN stay at the same network segment. NAT: WAN and LAN is different network segment, LAN port works as DHCP server and provide IP address.
LAN IP Address	Default LAN port IP is 192.168.100.1. IP length is 15 digits.
LAN MAC ID	Display LAN port MAC ID address.
DHCP Server Active	Default is Enable. ATA will assign IP address to DHCP clients at LAN port.
Assign IP	Default is 150~200. Define IP address range to DHCP clients. The range is 1~254. IP address format is 3 digits.
Lease Time	Default is 1:0 (day: hour). Define the DHCP IP lease time. The range is 0:0~ 12:23. Data length is 2 digits.
Submit [button]	Save the configuration.
Reset [button]	Clear the configuration.

3.4.2 DMZ & MAC Clone

Provide DMZ and MAC Clone configuration.



Field	Description
DMZ Type	Default is Disable. When set to Enable, all network packages will be sent to the IP address which was defined from [Assigned IP Address].
Assigned IP Address	Default is 192.168.123.150. IP format is: xxx.xxx.xxx.xxx. Data length is 15 digits.
MAC Clone Type	Default is Disable. When it was set to Enable, ATA will get your PC computer's MAC address. Option: Disable, Enable. ***Note:** 1. When use MAC Clone function, ATA must set ''LAN Mode" to NAT mode and enable DHCP server. 2. You have to login ATA to configure via LAN port to set the MAC Clone function.
Submit [button]	Save the configuration.
Reset [button]	Clear the configuration.

*** Note:**
If you want to resume your original MAC address, please use "**Restore Default Setting**" command at Webpage configuration.

3.4.3 Virtual Server

Provide 12 sets of Virtual Server.

Virtual Server Setting

Index	Active	Protocol	Internet Port Start ~ End	Extranet Port Start ~ End	Server IP Address	Action
1	☐	TCP	~	~		Delete
2	☐	TCP	~	~		Delete
3	☐	TCP	~	~		Delete
4	☐	TCP	~	~		Delete
5	☐	TCP	~	~		Delete
6	☐	TCP	~	~		Delete
7	☐	TCP	~	~		Delete
8	☐	TCP	~	~		Delete
9	☐	TCP	~	~		Delete
10	☐	TCP	~	~		Delete
11	☐	TCP	~	~		Delete
12	☐	TCP	~	~		Delete

Submit Reset Delete All

Field	Description
Index	Index number to support 12 sets configuration.
Enable	Default is Disable to all sets. When setting Enable, this function will be started.
Protocol	Default is TCP. Protocol option is: TCP or UDP.
Internet Port	Define the intranet port. Range is 1~65533. Here can define a range of ports or fixed port.
Extranet Port	Define the extranet port. Range is 1~65533.Here can define a range of ports or fixed port.
Server IP	Define internet server IP address which can only be IP address and the format is xxx.xxx.xxx.xxx. The address range is 15 digits.
Action	Click the delete button to clear the setting.
Submit [button]	Save the configuration.
Reset [button]	Clear the configuration.

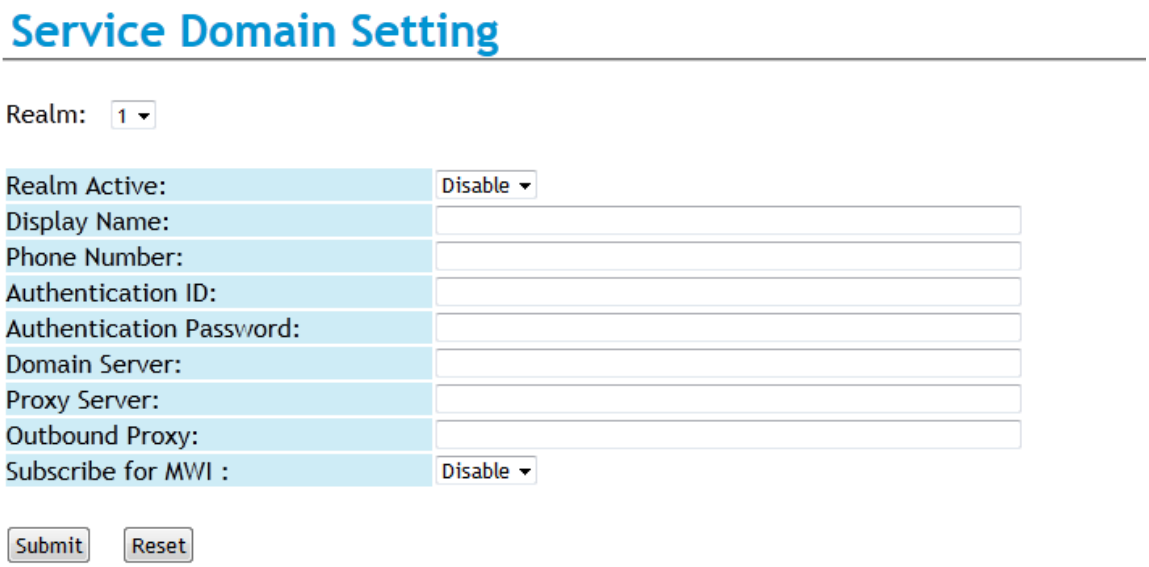
***Note:**
When you define the Internet & Extranet Ports, please avoid from using ATA default ports. For example: 5060, 9999, 20000.

3.5 SIP

Provide Service, Codes, Advanced and STUN functions.

3.5.1 Service

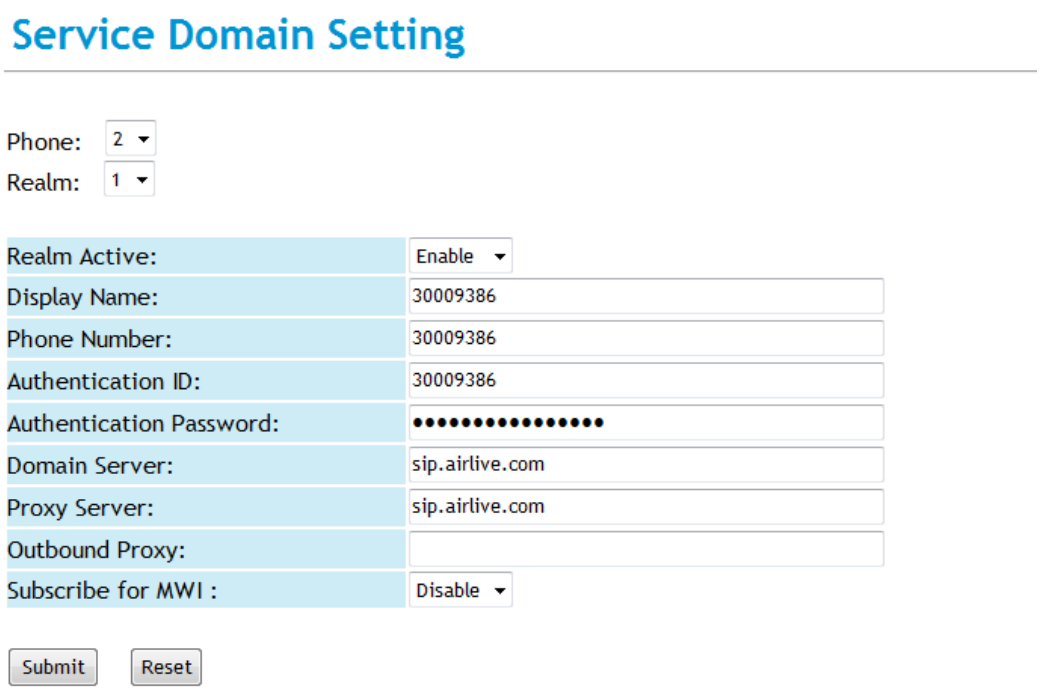
Provide 5 register accounts.



This figure is for VOIP-111A

Field	Description
Realm No.	Default is 1 (The first register account). If you want to switch to 1st account, please dial [1*] then hang up phone. Please refer to [Phone – Dial Plan Setting] -> [Realm 1~5 prefix].
Active	Default is Disable. This account will be active when you set to enable.
Display Name	Account's display name which can be numbers or strings. Maximum length is: 31 digits.
Phone Number	Account's phone number which only can be numbers. Maximum length is: 31 digits.
Authentication ID	Account's register ID which can be numbers or strings. Maximum length: 47 digits.
Authentication Password	Account's register password which can be numbers or strings. Maximum length is: 31 digits.
Domain Server	Input Domain Server. It can be IP Address or Domain Name. Format: xxx.xxx.xxx.xxx. Maximum length is 63 digits. If special Port Address is needed, please add it behind. For Example: nat.voiptalk.org:5065.

Proxy Server	Enter Proxy Server information. It can be IP Address or Domain Name. Format: xxx.xxx.xxx.xxx. Maximum length is 63 digits. If special Port Address is needed, please add it behind. For instance: nat.voiptalk.org:5065.
Outbound Proxy	Enter outbound Proxy Server information which can be IP Address or Domain Name. Format is: xxx.xxx.xxx.xxx. Maximum length is 63 digits. If special Port Address is needed, please add it behind. For instance: nat.voiptalk.org:5065.
Subscribe of MWI	Subscribe for MWI function (message waiting indicator). Your Register SIP Proxy server must support this function.
Submit [button]	Save the configuration.
Reset [button]	Clear the configuration.



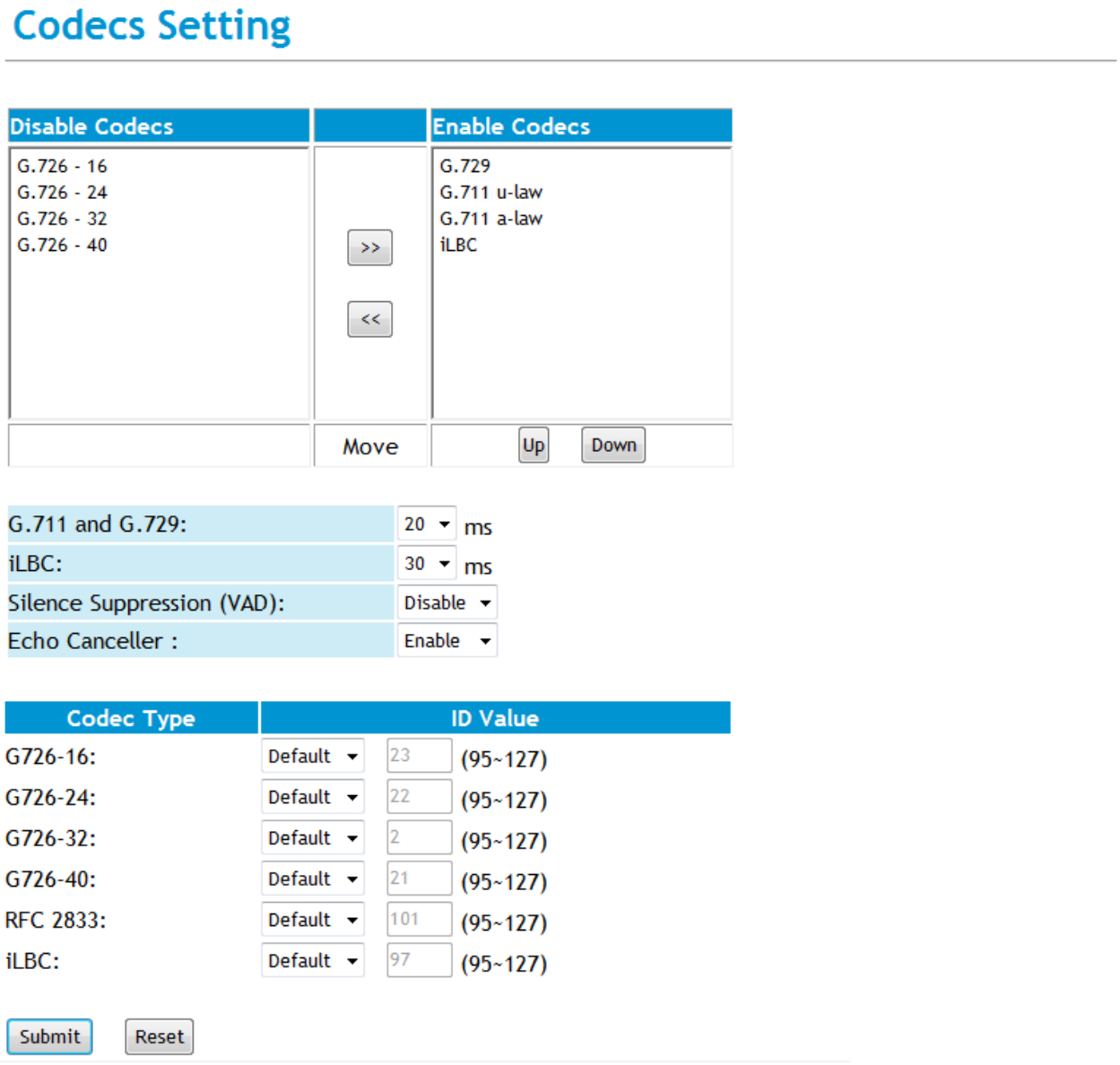
This figure is for VOIP-120A

Field	Description
Phone No	Default is Phone 1. Define phone 1~2 configuration.
Realm No.	Default is 1 (The first register account).If you want to switch to 1st account, please dial [1*] then hang up phone. Please refer to [Phone – Dial Plan Setting] -> [Realm 1~5 prefix].
Active	Default is Disable. This account will be active when you set to enable.
Display Name	Account's display name which can be numbers or strings. Maximum length: 31 digits.
Phone Number	Account's phone number which can only be numbers. Maximum length: 31 digits.
Authentication	Account's register ID which can be numbers or strings. Maximum

Field	Description
ID	length: 47 digits.
Authentication Password	Account's register password which can be numbers or strings. Maximum length: 31 digits.
Domain Server	Enter Domain Server which can be IP Address or Domain Name. Format is: xxx.xxx.xxx.xxx. Maximum length is 63 digits. If special Port Address is needed, please add it behind. For Example: nat.voiptalk.org:5065.
Proxy Server	Enter Proxy Server information which can be IP Address or Domain Name. Format is: xxx.xxx.xxx.xxx. Maximum length is 63 digits. If special Port Address is needed, please add it behind. For instance, nat.voiptalk.org:5065.
Outbound Proxy	Enter outbound Proxy Server information which can be IP Address or Domain Name. Format is: xxx.xxx.xxx.xxx. Maximum length is 63 digits. If special Port Address is needed, please add it behind. For instance, nat.voiptalk.org:5065.
Subscribe of MWI	Subscribe for MWI function (message waiting indicator). *Note: Your Register SIP Proxy server must support this function.
Submit [button]	Save the configuration.
Reset [button]	Clear the configuration.

3.5.2 Codes

Provide Voice Codec priority, RTP payload type and Codec ID setting.



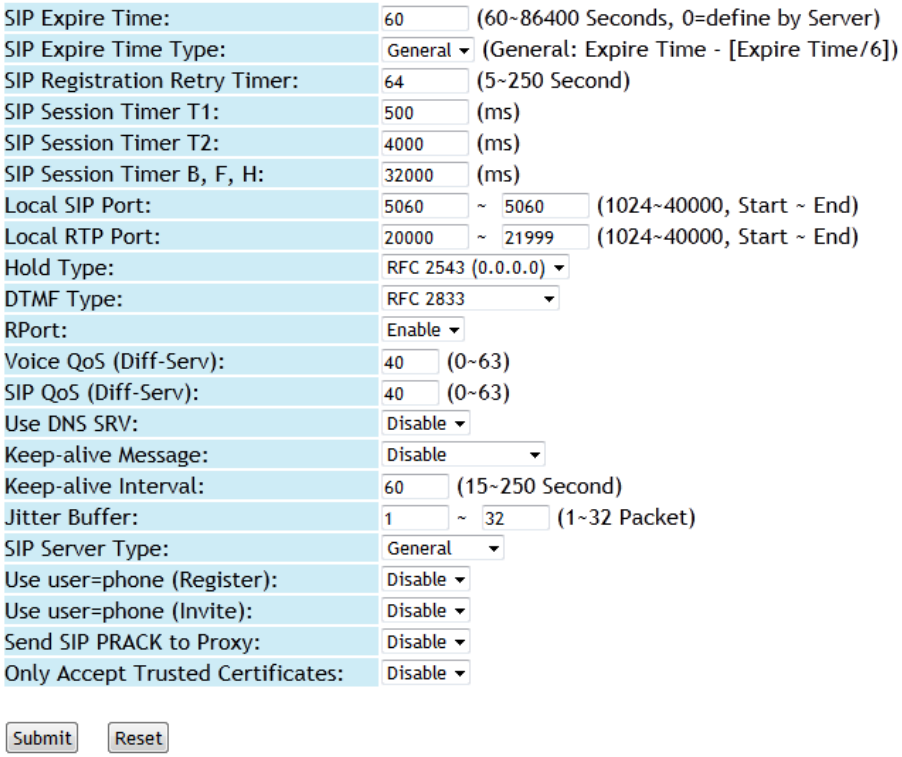
Field	Description
Disabled Codes	Default is G.726.16, G.726.24, G.726.32, G.726.40, iLBC. Provide disable codec item here.
>> **<<**	>>: move to Enable Codec <<: move to Disable Codec
Enabled Codec	Default is G.711 u-law, G.711 a-law, G.729 and G.723. Provide enable codec item here. The upper position codec has higher priority than lower one in this column.
G.711 and G.729	Default is 20 ms which defines G.711 and G.729 RTP payload type. Provide options are: 10, 20, 30, 40, 50, 60, 70, 80, and 90ms.
G.723	Default is 30 ms which defines G.723 RTP payload type. Provide

Field	Description
	options are: 30, 60, 90 ms.
G.723 5.3K	Default is Disable. Define G.723 5.3K function. When it was selected to enable the 5.3K function. Provide option: Disable, Enable.
Silence Suppression (VAD)	Default is Disable; When you set it to Enable, the (Voice Active Detection, VAD) function will be activated.The Silence Suppression is used in telephony to describe the process of not transmitting information over the network when one of the parties involved in a telephone call is not speaking, thereby reducing bandwidth usage.
Echo Cancel	Default is Disable. Enable: Enable the echo cancellation feature.
Codec Type	Define Voice Codec ID.
G726-16 ID	Default is 23; When you select “Customer Mode” (Manual change), the ID can be modified. (Range : 95~127)
G726-24 ID	Default is 22; When you select “Customer Mode” (Manual change), the ID can be modified. (Range : 95~127)
G726-32 ID	Default is 2; When you select “Customer Mode” (Manual change), the ID can be modified. (Range : 95~127)
G726-40 ID	Default is 21; When you select “Customer Mode” (Manual change), the ID can be modified. (Range : 95~127)
RFC 2833 ID	Default is 101; When you select “Customer Mode” (Manual change), the ID can be modified. (Range : 95~127)
Submit [button]	Save the configuration.
Reset [button]	Clear the configuration.

3.5.3 Advanced

Provide SIP Expire Time, SIP/RTP Port, QoS, Register SIP Expire Time, Use DNS SRV, DTMF, User=Phone, PRACK Function.

SIP - Advanced Setting


SIP Expire Time: 60 (60~86400 Seconds, 0=define by Server)
SIP Expire Time Type: General (General: Expire Time - [Expire Time/6])
SIP Registration Retry Timer: 64 (5~250 Second)
SIP Session Timer T1: 500 (ms)
SIP Session Timer T2: 4000 (ms)
SIP Session Timer B, F, H: 32000 (ms)
Local SIP Port: 5060 ~ 5060 (1024~40000, Start ~ End)
Local RTP Port: 20000 ~ 21999 (1024~40000, Start ~ End)
Hold Type: RFC 2543 (0.0.0.0)
DTMF Type: RFC 2833
RPort: Enable
Voice QoS (Diff-Serv): 40 (0~63)
SIP QoS (Diff-Serv): 40 (0~63)
Use DNS SRV: Disable
Keep-alive Message: Disable
Keep-alive Interval: 60 (15~250 Second)
Jitter Buffer: 1 ~ 32 (1~32 Packet)
SIP Server Type: General
Use user=phone (Register): Disable
Use user=phone (Invite): Disable
Send SIP PRACK to Proxy: Disable
Only Accept Trusted Certificates: Disable
Submit
Reset

This figure is for VOIP-111A

Field	Description
SIP Expire Time	Default is 60. The range is 60~86400 seconds. When it was set tp 0, ATA define expired time according to SIP Server. Data length is 5 digits.
SIP Expire Time Mode	Default is General. Option is: General, 1/2, 2/3, 3/4, 4/5, 5/6, 6/7, 7/8, 8/9, 9/10. This feature MUST be compatible with SIP Server. Calculate method: General: expire time-[(expire time/30) x6] and this time is longer than 60 seconds. If NOT, the time has to reduce 5 seconds as final time. 1/2: expire time x 1/2. 2/3: expire time x 2/3. 3/4: expire time x 3/4. 4/5: expire time x 4/5. 5/6: expire time x 5/6. 6/7: expire time x 6/7. 7/8: expire time x 7/8. 8/9: expire time x 8/9. 9/10: expire time x 9/10.
SIP Register Retry Interval	Default 64 seconds. Register Retry time when register fail. Range: 5~3600 seconds. Data length is 4 digits.

SIP T1	Default is 500 ms. Configure round-trip time (RTP) estimate. T1 determines several timers as defined in RFC3261. For example, when an unreliable transport protocol is used, a Client Invite transaction retransmits requests at an interval that start at T1 seconds and doubles after every retransmission. A Client General transaction retransmits requests at an interval that starts at T1 and doubles until it reaches T2. Range: 500~2000ms. Data length is 4 digits.
SIP T2	Default is 4000 ms. Determines the maximum retransmission interval for non-INVITE requests and INVITE responses. Range: 4000~16000 ms.
SIP Timer B, F, H	Default is 32000 ms. The maximum retransmit interval for non-INVITE requests and INVITE responses. Range: 8000~127000. Data length is 6 digits. B: 64 x SIP T1; INVITE transaction timeout timer. F: 64 x SIP T1; non-INVITE transaction timeout timer. H: 64 x SIP T1, Wait time for ACK receipt.
SIP Port Range of phone 1	Default is 5060~5060; Define Phone 1 SIP local port start and end port. Range: 1024~40000, ***Note:** 1. Both fields can be same or a range. 2. SIP port was used by two continuous ports. If SIP port was entered at 5060, two ports at 5060 and 5061 were used in reality. 3. All ports are unique and can not be duplicate.
RTP Port Range of phone 1	Default is 20000~20000; define Phone 1 RTP port start and stop. Range: 1024~40000. ***Note:** 1-Both fields can be same or a range. 2-RTP port was used by two continuous ports. If RTP port was entered at 20000, two ports at 20000 and 20001 were used in reality. 3-All ports are unique and can not be duplicate.
Hold by RFC	Default is 0.0.0.0. Configure to Hold on a call and define hold message sending method via SIP command. When it was Enable, [Connection Information (c): IN IP4 xxx.xxx.xxx.xxx] use IP address to set. Option: Disable, Type1 (Sendoloy), Type2 (inactvie).
DTMF Mode	Default is RFC 2833. Option: InBand, RFC2833, SIP Info, RFC2833 + Inband, SIP Info + Inband.
RPort	Default is Disable. When you set to [Enable], the [Message Header] will include [Rport] information.
Voice QoS (Diff-Serv)	Default is 40; Define Voice QoS (Diff-Ser) function. Range: 0~63.

SIP QoS (Diff-Serv)	Default is 40; Define SIP QoS (Diff-Ser) function. Range: 0~63.
Use DNS SRV	Default is Disable. When you enable the DNS SRV, ATA will run [DNS, Standard query SRV_sip_upd.xxx.xxx.xxx].
Send Keep Alives Packet	Default is Disable. If ATA was installed behind the NAT, it should keep the channel available so it has to send the keep alives packet to keep this channel. Enable: Use UDP format to send. For instance, UDP Source Port: sip, Destination Port:xxxx.
Keep Alives Period	Default is 60 seconds. Define the Keep Alives sending time period. Range: 15~250 seconds.
Jitter Buffer	Default is 1; Range: 0~32. A jitter buffer temporarily stores arriving voice packets in order to minimize delay variations to improve voice quality.
SIP Server Type	Default is General. Option: General, Asterisk, BroadWorks, Nortel, Xener, Vodtel, SKTelink.
Add URL user=phone (Register)	Default is Disable. ATA will put”user=phone” in register header when you set to enable.
Add URL user=phone (Invite)	Default is Disable. ATA will put”user=phone” in Invite header when you set to enable.
Send SIP PRACK of Proxy	Default is Disable. If you need to support PRACK, please enable this function.
Only Accept Call From Proxy	Default is Disable. Configure to accept incoming call which is registered to SIP Proxy server. Reject incoming call when use IP direct calling.
Submit [button]	Save the configuration.
Reset [button]	Clear the configuration.

SIP - Advanced Setting

SIP Expire Time:	60 (60~86400 Seconds, 0=define by Server)
SIP Expire Time Type:	General (General: Expire Time - [Expire Time/6])
SIP Registration Retry Timer:	64 (5~250 Second)
SIP Session Timer T1:	500 (ms)
SIP Session Timer T2:	4000 (ms)
SIP Session Timer B, F, H:	32000 (ms)
Local SIP Port of Phone 1:	5060 ~ 5060 (1024~40000, Start ~ End)
Local RTP Port of Phone 1:	20000 ~ 21999 (1024~40000, Start ~ End)
Local SIP Port of Phone 2:	5062 ~ 5062 (1024~40000, Start ~ End)
Local RTP Port of Phone 2:	22000 ~ 23999 (1024~40000, Start ~ End)
Hold Type:	RFC 2543 (0.0.0.0)
DTMF Type:	RFC 2833
RPort:	Enable
Voice QoS (Diff-Serv):	40 (0~63)
SIP QoS (Diff-Serv):	40 (0~63)
Use DNS SRV:	Disable
Keep-alive Message:	Disable
Keep-alive Interval:	60 (15~250 Second)
Jitter Buffer:	1 ~ 32 (1~32 Packet)
SIP Server Type:	General
Use user=phone (Register):	Disable
Use user=phone (Invite):	Disable
Send SIP PRACK to Proxy:	Disable
Only Accept Trusted Certificates:	Disable

Submit Reset

This figure is for VOIP-120A

Field	Description
SIP Expire Time	Default is 60. The range is 60~86400 seconds. When it was set tp 0, ATA define expired time according to SIP Server. Data length is 5 digits.

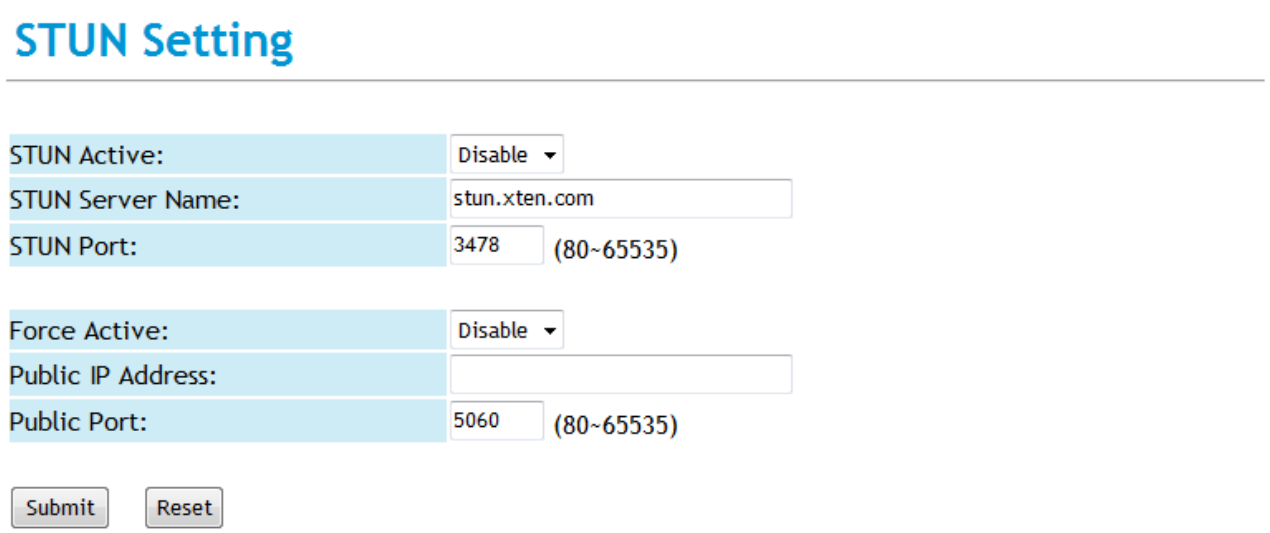
SIP Expire Time Mode	Default is General. Option is: General, 1/2, 2/3, 3/4, 4/5, 5/6, 6/7, 7/8, 8/9, 9/10. This feature MUST be compatible with SIP Server. Calculate method: General: expire time-[(expire time/30)x6] and this time is longer than 60 seconds. If NOT, the time has to reduce 5 seconds as final time. 1/2: expire time x 1/2. 2/3: expire time x 2/3. 3/4: expire time x 3/4. 4/5: expire time x 4/5. 5/6: expire time x 5/6. 6/7: expire time x 6/7. 7/8: expire time x 7/8. 8/9: expire time x 8/9. 9/10: expire time x 9/10.
SIP Register Retry Interval	Default 64 seconds. Register Retry time when register fail. Range: 5~3600 seconds. Data length is 4 digits.
SIP T1	Default is 500 ms. Configure round-trip time (RTP) estimate. T1 determines several timers as defined in RFC3261. For example, when an unreliable transport protocol is used, a Client Invite transaction retransmits requests at an interval that start at T1 seconds and doubles after every retransmission. A Client General transaction retransmits requests at an interval that starts at T1 and doubles until it reaches T2. Range: 500~2000ms. Data length is 4 digits.
SIP T2	Default is 4000 ms. Determines the maximum retransmission interval for non-INVITE requests and INVITE responses. Range: 4000~16000 ms.
SIP Timer B, F, H	Default is 32000 ms. The maximum retransmit interval for non-INVITE requests and INVITE responses. Range: 8000~127000. Data length is 6 digits. B: 64 x SIP T1; INVITE transaction timeout timer. F: 64 x SIP T1; non-INVITE transaction timeout timer. H: 64 x SIP T1, Wait time for ACK receipt.
SIP Port Range of phone 1	Default is 5060~5060; Define Phone 1 SIP local port start and end port. Range: 1024~40000, ***Note:** 1- Both fields can be same or a range. 2- SIP port was used by two continuous ports. If SIP port was entered at 5060, two ports at 5060 and 5061 were used in reality. 3-All ports are unique and can not be duplicated.

RTP Port Range of phone 1	Default is 20000~20000; define Phone 1 RTP port start and stop. Range: 1024~40000. ***Note:** 1-Both fields can be same or a range. 2-RTP port was used by two continuous ports. If RTP port was entered at 20000, two ports at 20000 and 20001 were used in reality. 3-All ports are unique and can not be duplicated.
SIP Port Range of phone 2	Default is 5062~5062; Define Phone 2 SIP local port start and end port. Range: 1024~40000, ***Note:** 1- Both fields can be same or a range. 2- SIP port was used by two continuous ports. If SIP port was entered at 5062, two ports at 5062 and 5063 were used in reality. 3-All ports are unique and can not be duplicate.
RTP Port Range of phone 2	Default is 20002~20002; define Phone 2 RTP port start and stop. Range: 1024~40000. ***Note:** 1-Both fields can be same or a range. 2-RTP port was used by two continuous ports. If RTP port was entered at 20002, two ports at 20002 and 20003 were used in reality. 3-All ports are unique and can not be duplicate.
Hold by RFC	Default is 0.0.0.0. Configure to Hold on a call and define hold message sending method via SIP command. When it was Enable, [Connection Information (c): IN IP4 xxx.xxx.xxx.xxx] use IP address to set. Option: Disable, Type1 (Sendoloy), Type2 (inactvie).
DTMF Mode	Default is RFC 2833. Option: InBand, RFC2833, SIP Info, RFC2833 + Inband, SIP Info + Inband.
RPort	Default is Disable. When you set to [Enable], the [Message Header] will include [Rport] information.
Voice QoS (Diff-Serv)	Default is 40; Define Voice QoS (Diff-Ser) function. Range: 0~63.
SIP QoS (Diff-Serv)	Default is 40; Define SIP QoS (Diff-Ser) function. Range: 0~63.
Use DNS SRV	Default is Disable . When you enable the DNS SRV, ATA will run [DNS, Standard query SRV_sip_upd.xxx.xxx.xxx].
Send Keep Alives Packet	Default is Disable. If ATA was installed behind the NAT, it should keep the channel available so it has to send the keep alives packet to keep this channel. Enable: Use UDP format to send. For instance, UDP Source Port: sip, Destination Port:xxxx.

Keep Alives Period	Default is 60 seconds. Define the Keep Alive sending time period. Range: 15~250 seconds.
Jitter Buffer	Default is 1; Range: 0~32. A jitter buffer temporarily stores arriving voice packets in order to minimize delay variations to improve voice quality.
SIP Server Type	Default is General. Option: General, Asterisk, BroadWorks, Nortel, Xener, Vodtel, SKTelink.
Add URL user=phone (Register)	Default is Disable. ATA will put"user=phone" in register header when you set to enable.
Add URL user=phone (Invite)	Default is Disable. ATA will put" user=phone" in Invite header when you set to enable.
Send SIP PRACK of Proxy	Default is Disable. If you need support PRACK, please enable this function.
Only Accept Call From Proxy	Default is Disable. Configure to accept incoming call which is registered to SIP Proxy server. Reject incoming call when use IP direct calling.
Submit [button]	Save the configuration.
Reset [button]	Clear the configuration.

3.5.4 STUN

Provide STUN and Force setting.



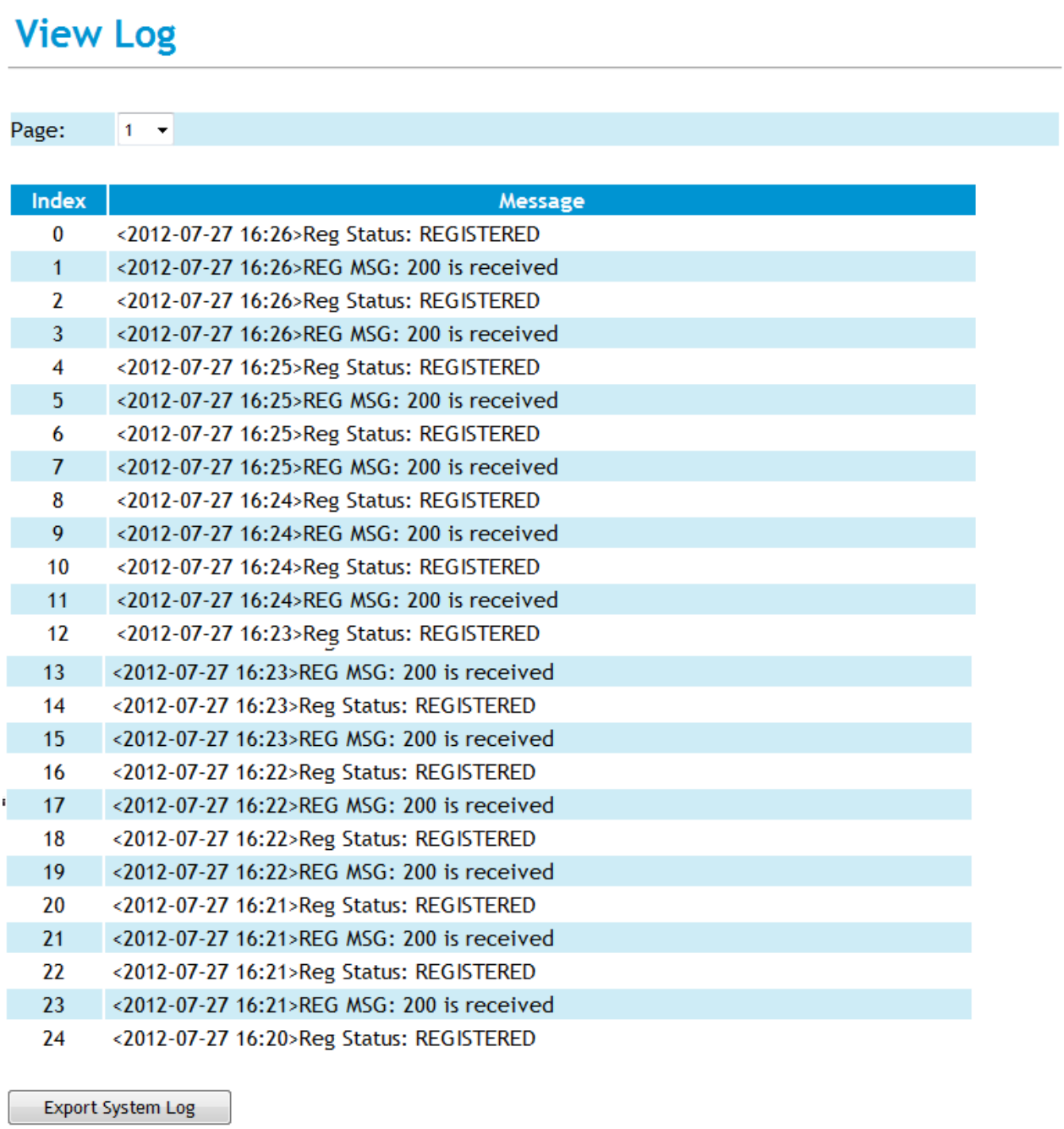
Field	Description
STUN Active	Default is Disable. Option: Disable, Enable.
STUN Server Name	Default is stun.xten.com; Define STUN server IP which can be IP address or Domain Name. The format is xxx.xxx.xxx.xxx. Data length is 63 digits.

STUN Port Number	Default port is 3478; Define STUN port. Range: 80~65535. Data length is 5 digits.
Force Active	Default is Disable. When this feature was enabled, ATA change SIP messge to IP. Option: Disable, Enable.
Public IP Address	Define Router external IP address. Data length is 63 digits. The format is xxx.xxx.xxx.xxx.
Public Port Number	Default port is 3478. Define Router external port number. Range: 80~65535.
Submit [button]	Save the configuration.
Reset [button]	Clear the configuration.

3.6 Management (Advanced Configuration)

Provide Status Log, Auto Config, Auto Update, New Firmware, Advanced, Password, Tones, Default, Language functions.

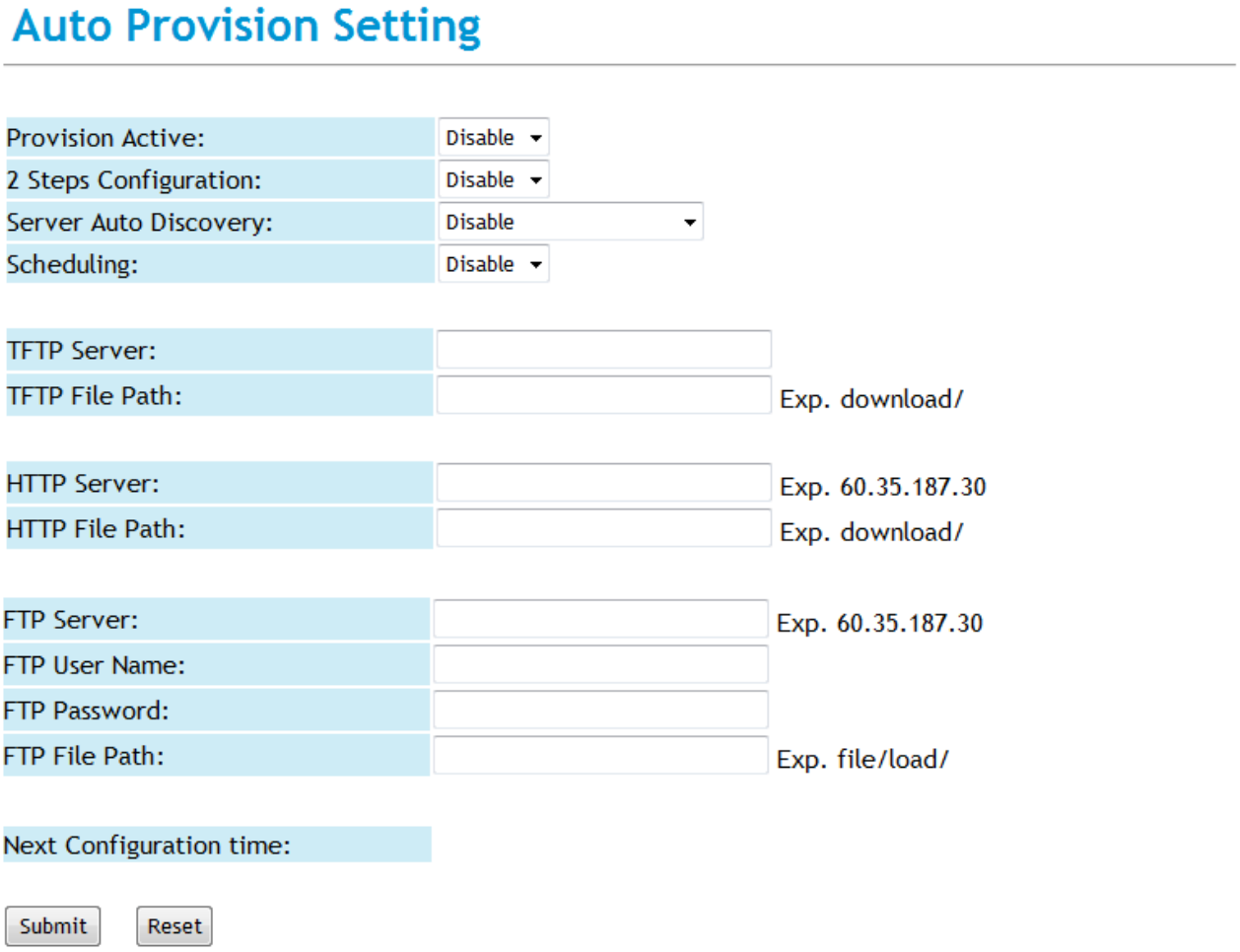
3.6.1 View Log



Field	Description
Page	Default is 1(page 1).
Index	Information ID.
Content	Include time and information. <2010-09-14 10:35> Get Time from SNTP server, Succeed! <2010-09-14 10:35>: event time and date. Get Time from SNTP server, Succeed!: event
Get Status Log [button]	Save the [Status Log] to file; default file name is Syslog.log.

3.6.2 Auto Provision

Provide 3 kinds of provision methods. (TFTP, FTP and HTTP)

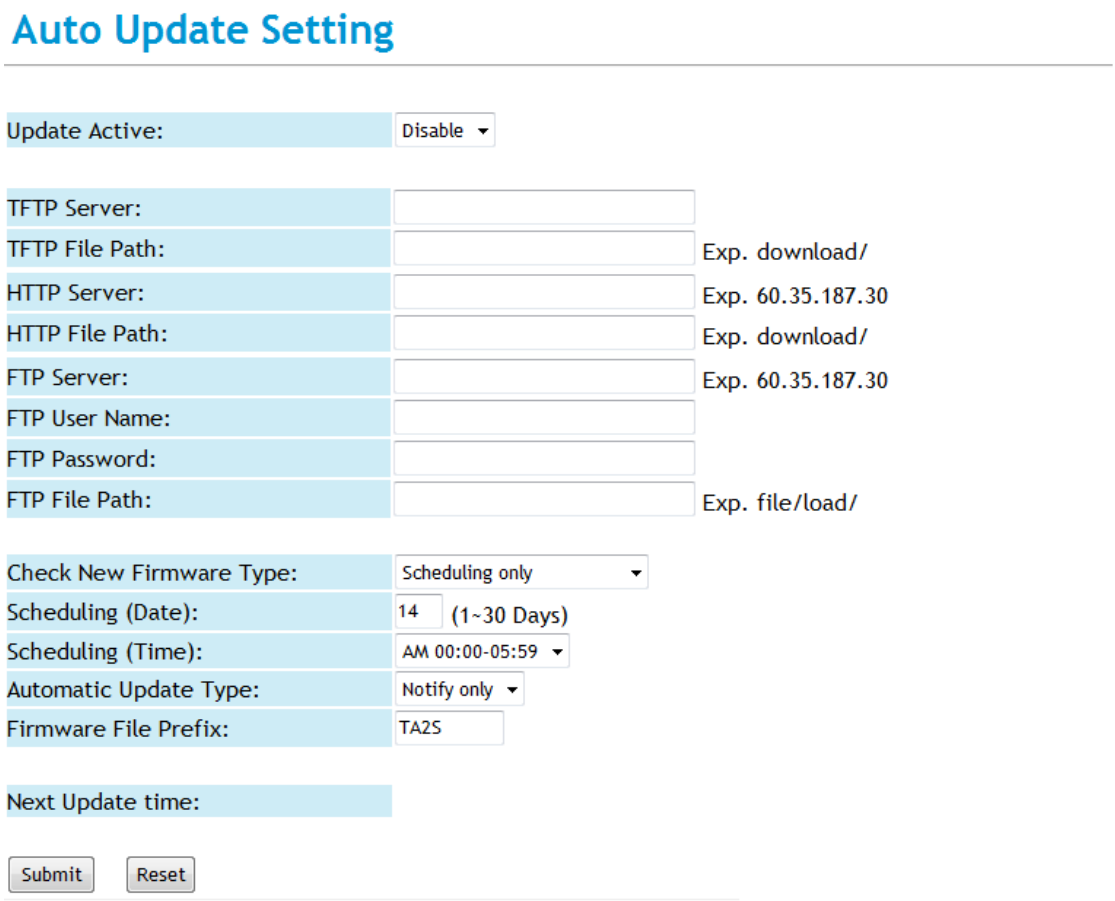


Field	Description
Type	Default is Disable; Define auto configure method. When it was enabled, ATA visit assigned Server to download “MACID.dat” file. Option: Disable, TFTP, FTP and HTTP.
2 Steps configuration	Default is Disable; Define 2 stages update process. The first step is to download common used messages. The second step is to SIP Server registration Account and Password. Option: Disable , Enable.
Server auto discover	Default is Disable; Define discover provision server method. Option: Disable, DHCP TFTP Option 66 (TFTP), Broadcasting. DHCP TFTP Option 66 (TFTP): When DHCP server assigns one IP to ATA, it also sends the Option 66 server IP to ATA. And ATA will fill in the Optoin 66 IP to [TFTP Server] field. Broadcasting: Using broadcasting method to search for Server.
Scheduling	Default is Disable; When you enable the Scheduling, ATA will follow the [Next Config Time] to check or update configuration data. Option: Disable, Enable.
TFTP Server	Define TFTP server IP. The format is xxx.xxx.xxx.xxx. Data length is 15 digits.

Field	Description
TFTP File Path	TFTP configuration files localtion. It can be numbers or strings. Maximum length: 63 digits. For Instance: 123/ to setup file folder. An "/" has to enter behind.
HTTP Server	Define the HTTP Address which can be IP Address or Domain Name. Format: xxx.xxx.xxx.xxx; Maximum length: 63 digits.
HTTP File Path	HTTP configuration files location. It can be numbers or strings. Maximum length: 63 digits. For Instance: 123/ to setup file folder. An "/" has to enter behind.
FTP Server	Define the FTP Address which can be IP Address or Domain Name. Format: xxx.xxx.xxx.xxx; Maximum length: 63 digits.
FTP Username	FTP server user name. Data length is 63 digits.
FTP Password	FTP server user password. Data length is 63 digits.
FTP file Path	FTP configuration files location. It can be numbers or strings. Maximum length: 63 digits. For Instance: 123/ to setup file folder. An "/" has to enter behind.
Next config time	Display the next configuration date and time. If the estimated date was expired, ATA will visit assigned server to check the updated information.
Submit [button]	Save the configuration.
Reset [button]	Clear the configuration.

3.6.3 Auto Update

Provide **rom** format file upgrade.



Field	Description
Type	Default is Disable ; Define version upgrade method. Option: Disable, TFTP, FTP or HTTP mode.
TFTP Server	Set up TFTP Server IP address which can be IP Address or Domain Name. Format: xxx.xxx.xxx.xxx. Data length is 15 digits.
TFTP Path	Set up TFTP Path. Enter the path of the file which can be numbers or strings. Maximum length is 63 digits. For Instance: 123/ to setup file folder. An "/" has to enter behind.
HTTP Server	Define the HTTP Address which can be IP Address or Domain Name. Format: xxx.xxx.xxx.xxx; Maximum length: 63 digits.
HTTP Path	HTTP configuration files location. It can be numbers or strings. Maximum length: 63 digits. For Instance: 123/ to setup file folder. An "/" has to enter behind.
FTP Server	Define the FTP Address which can be IP Address or Domain Name. Format: xxx.xxx.xxx.xxx; Maximum length: 63 digits.
FTP Username	FTP server user name. Data length is 63 digits.
FTP Password	Set up FTP login password. Data length is 63 digits.
File Path	FTP configuration files location. It can be numbers or strings. Maximum length: 63 digits. For Instance: 123/ to setup file folder. An "/" has to enter behind.
Check new Firmware Type	Default is Scheduling Only. Option: Power on and Schedule, Scheduling Only. - Power on and Scheduling: When the ATA boot up or reach the schedule time, it will check if there is any upgraded firmware. - Scheduling: ATA will follow the [Next Update Time] to check if there is any new version for the upgrade. * Power on and Scheduling: ATA doesn't upgrade firmware automatically when it finds new version. ATA will prompt an alert tone to user. And User must upgrade it manually.
Scheduling (Date)	Default is 14 day; Define the scheduling date. Minimum: 1 day. Maximum: 30 days. Only numbers are accepted, length: 2 digits.
Scheduling (Time)	Default: AM 00:00 – 05:59、AM 00:00 – 05:59、AM 06:00 – 11:59、AM 12:00 – 17:59、AM 18:00 – 23:59 is available.
Automatic Update	Default is Notify only. Option: Notify only, Automatic - Notify only: ATA will not auto update when it found new version. ATA will prompt an alert tone to user. - Automatic: ATA will auto update when it finds new version.
Firmware File Prefix	Default: Product model. Can be numbers or strings, maximum: 8 digits.
Next Update Time	Showing the next update time. ***Note:** Date calculation starts from next day**.**
Submit [button]	Save the configuration.
Reset [button]	Clear the configuration.

***Note:** Firmware updated manually at Auto Provision mode.

1. You will hear “DuDuDu” alert tone from handset when you pickup handset after updated firmware was available. If you give up update firmware procedure here, you don’t hear any alert tone at next time.
2. If you want to proceed firmware upgrade procedures, dial #190# and hang on handset.
3. Pick up Handset again; dial #160# to enter firmware upgrade procedures.

Once ATA enter firmware upgrade procedures, it takes about 2 to 3 minutes to complete. ATA don’t implement any job or function in this period. Please don’t unplug power adaptor during firmware upgrade procedures in order to prevent from failure.

3.6.4 Firmware

The Firmware upgrade only support rom format.

Field	Description
Code Type	xxx.rom and xxxx.ssh
File Location	Update firmware location. Data length is 30 digits.
Update	Starting upgrade firmware.
Reset	Clear the file location information.

***Note:** It takes 2 or 3 minutes when you start update firmware. User can’t make call during the upgrade. Please don’t turn off the power.

VOIP-111’s firmware are different with VOIP-120A, please only use correct firmware for these models.

3.6.5 Advanced

Provide anonymous call, billing signal, encryption, syslog and FXS/FXO parameter function setting.

Management - Advanced Setting

ICMP Not Echo:	Disable
Anonymous Call:	Disable
Management from WAN:	Enable
Stop Feature Tone:	Disable (MWI, forward, Do Not Disturb....)
Billing Signal:	Disable
CPC Delay:	2 (seconds)
CPC Duration:	0 (0~120; x 10ms)
IP Dialing Format:	Type 1 (x@x.x.x.x)
Send Flash Event:	Disable
Encryption Type:	Disable
Encryption Key:	
PPPoE Retry Period:	5 (0~250 seconds)
DHCP Gateway ARP Check Period:	0 (0 or 30~300 seconds)
Syslog Server IP Address:	
System Log:	Disable
FXS Port Country:	USA
Flash Hook Time (MAX):	60 (4~255; x 10ms)
Flash Hook Time (MIN):	7 (3~12; x 10ms)
NET Bandwidth Limit:	Disable Kbps

Submit Reset

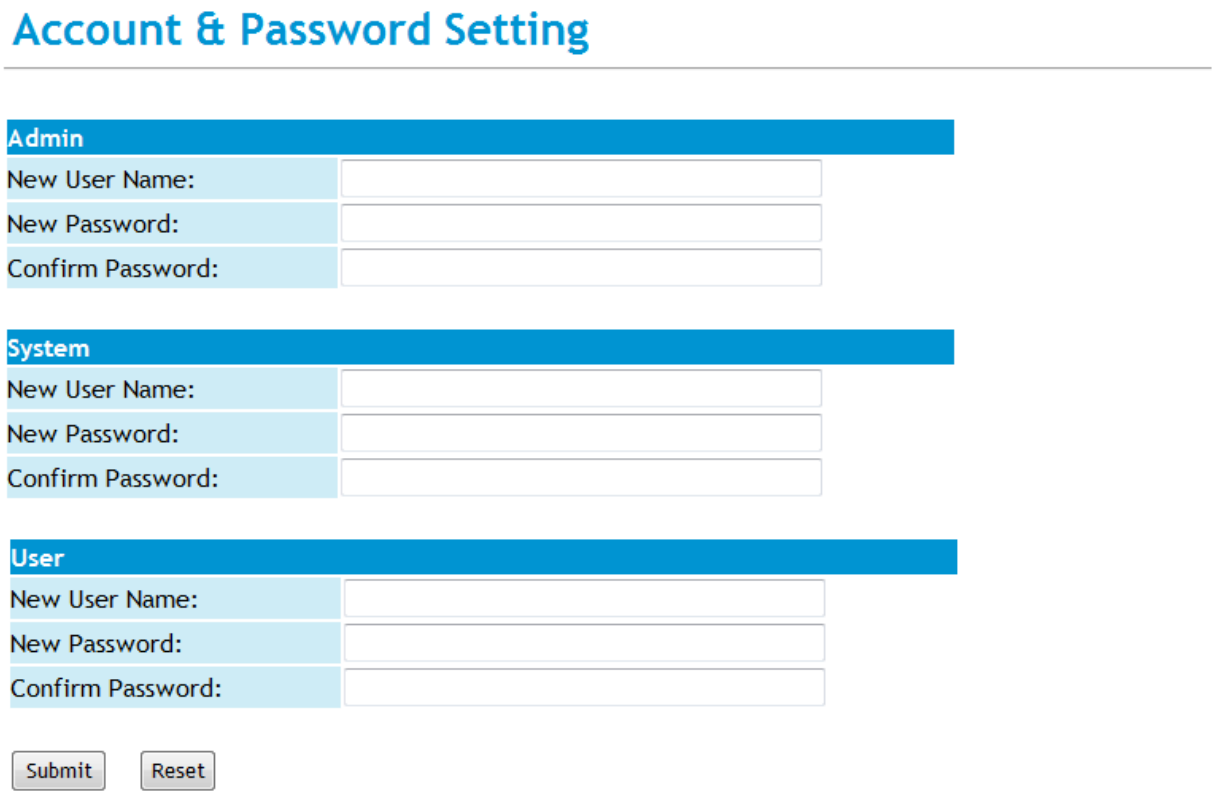
Field	Description
ICMP Not Echo	Default is Disable. When ICMP was set to Enable, ATA doesn't response PING command. Option: Disable, Enable
Send Anonymous CID	Default is Disable. When ATA was set to Type 1 or Type 2, ATA will send out anonymous to SIP Server (or to remote party) instead of CID. Type 1 (anonymous@anonymous.invalid) Type 2 (anonymous@x.x.x.x) ***Note:** Your Register Proxy server must support this function.
Management from WAN	Default is Enable which allows web management access from WAN port. When it was set to Disable, ATA only allow web access vial LAN port. Option: Disable, Enable

Stop Feature Tone	Default is Disable. This feature is to provide alert tone at the following messages. Enable: If you enable [Subscribe for MWI, forward, DND] function, you will hear the alert tone (DuDuDu…..) when you pick up the phone. Option: Disable, Enable.
Billing Signal	Default is Disable. This feature is to provide start billing signal when call was established. Option: Disable, Polarity Reversal, Tone_12K, Tone_16K.
CPC Delay	Default is 2 seconds. Delay how many seconds to send CPC signal (Loop Current Drop signal) to Analog telephone set when ATA received drop call signal from IP SIP command. Only numbers are accepted, data range is (2~5 seconds), maximum data length is 1 digit.
CPC Duration	Default is 120ms. Setting CPC feature was activated duration (Loop Current Drop duration), data range is (0~120ms), maximum length: 3 digits.
IP Dialing Format	Default is Type 1 (x@x.x.x.x); Define the IP dialing format. Option: Disabled, Type 1 (x@x.x.x.x.), Type 2 (x.x.x.x).
Send Flash event	Default is Disable; When you press Flash Key at analog phone set to do Transfer feature, ATA will send different event messages to IP side. Option: Disable: Send [SIP/DSP, Content-Type=applicatio-sdp]. DTMF Event: Send [RTP event, Payload type=RTP event Flash]. SIP Info: send [SIP, Request: INFO sip:xxx@xxxx].
Encrypt Type	Default is Disable. Option: Disable, INFINET, AVS, WALKERSUN1, WALKERSUN2, CSF1, CSF2, GX, VGX, RC4, VOS_R, VGCP and Welltech. VGCP is popular in the market. Once this featuare was selected, both voice codec and SIP command were encrypted during transmit on IP network. ***Note:** Your Registered Proxy server must support the same encryption type with ATA.
Encrypt Key	Set encryption password. Only VGX encryption format need password. Maximum data is 63 digits which can be numbers or strings.
PPPoE Retry Period	Default is 5 (Seconds). Set the time for PPPoE to retry when PPPoE failed. Only numbers are accepted, data range: (5~255) seconds, maximum length is 3 digits.
System Log Server	Sending ATA debug messages to System Log Server which can be IP Address or Domain Name Address. Format: xxx.xxx.xxx.xxx; Maximum lengthis 63 digits.

System Log Type	Default is Disable; Define Syslog type or Log message type. Option: Disable, Call Statistics, General Debug, Call Statistics + General Debug, SIP Debug, Call Statistics + SIP Debug, General Debug + SIP Debug, All.
FXS Port Country	Default is USA. To select FXS Port impedance of the analog telephone by different country's specification.
Flash Signal Detect (Max)	Default is 60 (equal 600ms). To detect Hook Flash Time at maximum time. When Flash time is less than 600ms, it was regarded as HOOK FLASH command. When Flash time is longer than 600ms, it was regarded as On-Hook (drop call) command. Configuration range is from (4~255), Unit: 10ms. Maximum length is 3 digits.
Flash Signal Detect (Min)	Default is 7(equal to 70ms). When Flash Time is longer than 70ms, it will be regarded as FLASH command. When Flash Time is less than 70 ms, it will be regarded as On-Hook. Configuration range is from (3~12), Unit: 10ms. Minimum length is 3 digits.
NET Bandwidth Limit	Default is Disable. LAN port bandwidth limitation. Option: Disable, 128, 256, 512, 1024, 2048, 4096, 8192 kbps.
Submit [button]	Save the configuration.
Reset [button]	Clear the configuration.

3.6.6 Password

There are 3 levels login name and password.



Field	Description
Admin	Administrator (The highest level): ATA only provide one administrator account. This level can configure all setting pages such as: [Phone: Phone Book, Dial Plan, Call Service, General, Volume; Network: WAN, DDNS, VLAN, VPN, SNTP; NAT: LAN, DMZ, Virtual Server; SIP: Service, Code, Advanced, Stun; Management: Status Log, Auto Config, Auto Update, New Firmware, Advanced, Password, Tones, Default, Language; Save & Reboot, Logout]. Default user name: **airlive**, default user password: **airlive**.
New username	Enter new username which can be Numbers or strings, maximum length is 32 digits.
New password	Enter new password which can be Numbers or strings, maximum length is 32 digits.
Confirmed password	Enter new password to confirm the password setting.
System	System user: ATA only provide one administrator account. This level can modify below setting, [Phone: Phone Book, Dial Plan, Call Service, General, Volume; Network: WAN, DDNS, VLAN, VPN, SNTP; NAT: LAN, DMZ, Virtual Server; SIP: Service, Code, Stun; Management: Status Log, Auto Config, Auto Update, New Firmware, Password, Default, Language; Save & Reboot, Logout]. Default user name: **system**, Default user password: **airlive**.
New username	Enter new username which can be Numbers or strings, maximum length is 32 digits.
New password	Enter new password which can be Numbers or strings, maximum length is 32 digits.
Confirmed password	Enter new password to confirm the password setting.
User	Normal User: ATA only provides one normal user account. This level only modify below setting, [Phone: Phone Book, Call Service, General, Volume; Network: WAN, DDNS, NAT: LAN, DMZ, Virtual Server; Management: Status Log, Language, Save & Reboot, Logout]. Default user name: **user**, Default user password: **airlive**.
New username	Enter new username which can be Numbers or strings, maximum length is 32 digits.
New password	Enter new password which can be Numbers or strings, maximum length is 32 digits.
Confirmed password	Enter new password to confirm the password setting.
Submit [button]	Save the configuration.
Reset [button]	Clear the configuration.

3.6.7 Tone

Tone setting provides Dial, Ring Back, Busy, Congestion, Ring, Call Waiting Tone and Multi-Frequency configuration.

Tones Setting

	Dial	Ring Back	Busy	Congestion	Ring	Call Waiting
Cadence On:	☐	☑	☑	☑	☑	☑
Hi-Tone Freq.:	440	480	620	620	480	440
Lo-Tone Freq.:	350	440	480	480	440	350
Hi-Tone Gain:	2261	2261	2261	2261	15360	2261
Lo-Tone Gain:	2261	2261	2261	2261	15360	1130
On Time 1:	0 x 10ms	200	50	30	200	30
Off Time 1:	0 x 10ms	400	50	20	400	20
On Time 2:	0 x 10ms	0	0	0	0	30
Off Time 2:	0 x 10ms	0	0	0	0	400
On Time 3:	0 x 10ms	0	0	0	0	0
Off Time 3:	0 x 10ms	0	0	0	0	0

Submit Reset

Tone Gain Value: 372767-> 0bB, 16384-> -6dB, 8192-> -12dB

Field	Description
Dial Tone	Dial tone configuration.
Cadence On	Default is disable.
Hi-Tone Freq	Default is 440. Only numerals are acceptable. Data range: (0~4096). Maximum length: 4 bytes.
Lo-Tone Freq	Default is 350. Only numerals are acceptable. Data range: (0~4096). Maximum length: 4 bytes.
Hi-Tone Gain	Default: 4522; Only numerals are acceptable. Data range: (0~65535). Maximum length: 5 bytes.
Lo-Tone Gain	Default: 2261; Only numerals are acceptable. Data range: (0~65535). Maximum length: 5 bytes.
On Time 1	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
Off Time 1	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
On Time 2	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
Off Time 2	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.

On Time 3	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
Off Time 3	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
Ring Back	Ring back tone configuration.
Candence On	Default is enable
Hi-Tone Freq	Default: 440; Only numerals are acceptable. Data range: (0~4096). Maximum length: 4 bytes.
Lo-Tone Freq	Default: 350; Only numerals are acceptable. Data range: (0~4096). Maximum length: 4 bytes.
Hi-Tone Gain	Default: 4522; Only numerals are acceptable. Data range: (0~65535). Maximum length: 5 bytes.
Lo-Tone Gain	Default: 2261; Only numerals are acceptable. Data range: (0~65535). Maximum length: 5 bytes.
On Time 1	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
Off Time 1	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
On Time 2	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
Off Time 2	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
On Time 3	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
Off Time 3	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
Busy	Busy Tone configuration.
Candence On	Default is enable
Hi-Tone Freq	Default: 440; Only numerals are acceptable. Data range: (0~4096). Maximum length: 4 bytes.
Lo-Tone Freq	Default: 350; Only numerals are acceptable. Data range: (0~4096). Maximum length: 4 bytes.
Hi-Tone Gain	Default: 4522; Only numerals are acceptable. Data range: (0~65535). Maximum length: 5 bytes.
Lo-Tone Gain	Default: 2261; Only numerals are acceptable. Data range: (0~65535). Maximum length: 5 bytes.
On Time 1	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
Off Time 1	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
On Time 2	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
Off Time 2	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
On Time 3	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.

Off Time 3	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
Congestion	Congestion tone configuration
Candence On	Default is enable.
Hi-Tone Freq	Default: 440; Only numerals are acceptable. Data range: (0~4096). Maximum length: 4 bytes.
Lo-Tone Freq	Default: 350; Only numerals are acceptable. Data range: (0~4096). Maximum length: 4 bytes.
Hi-Tone Gain	Default: 4522; Only numerals are acceptable. Data range: (0~65535). Maximum length: 5 bytes.
Lo-Tone Gain	Default: 2261; Only numerals are acceptable. Data range: (0~65535). Maximum length: 5 bytes.
On Time 1	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
Off Time 1	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
On Time 2	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
Off Time 2	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
On Time 3	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
Off Time 3	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
Ring	Ring tone configuration.
Candence On	Default is enable
Hi-Tone Freq	Default: 440; Only numerals are acceptable. Data range: (0~4096). Maximum length: 4 bytes.
Lo-Tone Freq	Default: 350; Only numerals are acceptable. Data range: (0~4096). Maximum length: 4 bytes.
Hi-Tone Gain	Default: 4522; Only numerals are acceptable. Data range: (0~65535). Maximum length: 5 bytes.
Lo-Tone Gain	Default: 2261; Only numerals are acceptable. Data range: (0~65535). Maximum length: 5 bytes.
On Time 1	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
Off Time 1	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
On Time 2	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
Off Time 2	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
On Time 3	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
Off Time 3	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.

Call Waiting	Call waiting tone configuration.
Candence On	Default is enable
Hi-Tone Freq	Default: 440; Only numerals are acceptable. Data range: (0~4096). Maximum length: 4 bytes.
Lo-Tone Freq	Default: 350; Only numerals are acceptable. Data range: (0~4096). Maximum length: 4 bytes.
Hi-Tone Gain	Default: 4522; Only numerals are acceptable. Data range: (0~65535). Maximum length: 5 bytes.
Lo-Tone Gain	Default: 2261; Only numerals are acceptable. Data range: (0~65535). Maximum length: 5 bytes.
On Time 1	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
Off Time 1	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
On Time 2	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
Off Time 2	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
On Time 3	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
Off Time 3	Default is 0; The unit is 10ms; Only numerals are acceptable. Data range: 0~ 999, Maximum length: 3 bytes.
Submit [button]	Save the configuration.
Reset [button]	Clear the configuration.

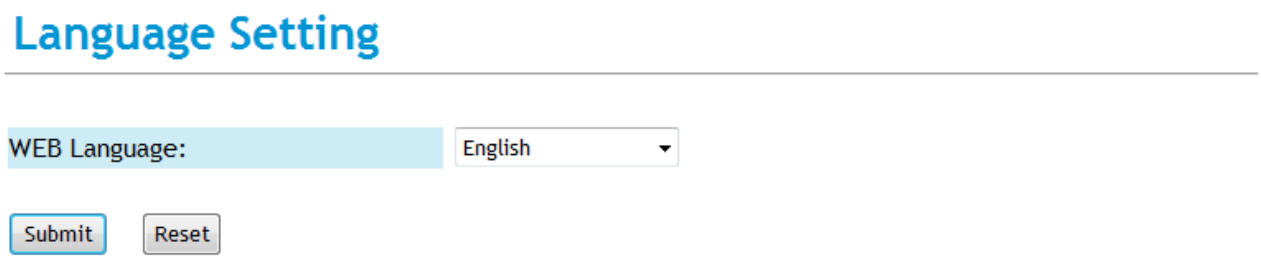
3.6.8 Default

Restore whole configuration setting to factory default except Phone Book.

Field	Description
Reset to Factory Setting	Restore to default setting and reboot.
Export Setting	Export the configure file.
Restore Setting	Restore configuration data which was downloaded via ATA.

3.6.9 Language

Provide language option for web configuration, ATA will auto reboot after press [Submit]. Don't remove Power adaptor at this moment.



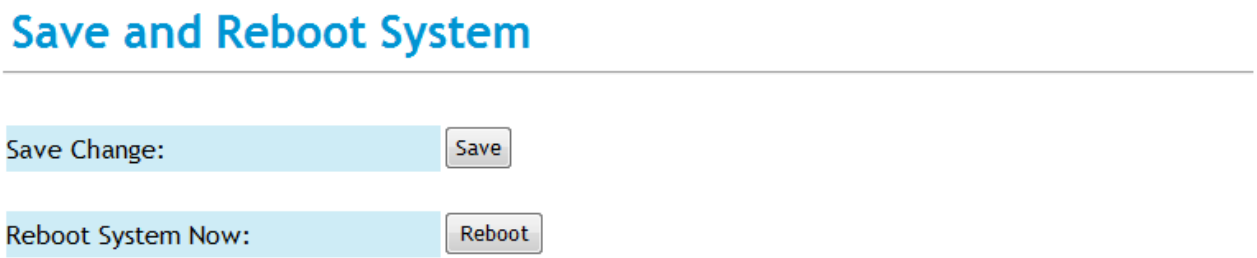
Field	Description
Choice Language	Default is English. ATA has to restart after you have changed Web language. Option: English, Chinese, Simplified Chinese, Portuguese.
Submit [button]	Save the configuration.
Reset [button]	Clear the configuration.

3.7 Save & Reboot

Save configuration and Reboot ATA.

3.7.1 Save & Reboot

- Save Change: Save configuration and auto reboot to take effect.
- Reboot System: Reboot ATA



Field	Description
Save	Save configuration and auto reboot.
Reboot	Reboot ATA.

3.8 Logout

3.8.1 Logout

Logout system and return to login page.

Field	Description
Logout	Logout system and return to login page.